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Contents of the Volume

From the Chief Author's Desk

The research activities among different disciplines of natural science are backbone of system. The deep and strong affords are the demands of today. Sincere afford must be exposed worldwide. Which, in turns, require international platform for rapid and proper communication among similar and interdisciplinary research groups.

The Global Journal of Computer Science and Technology is to fulfill all such demands and requirements, and functions also as an international platform. Of course, the publication of research work must be reviewed to establish its authenticity. This helps to promote research activity also. We know, great scientific research have been worked out by philosopher seeking to verify quite erroneous theories about the nature of things.

The research activities are increasing exponentially. These great increments require rapid communication, also to link up with others. The balanced communication among same and interdisciplinary research groups is major hurdle to aware with status of any research field.

The Global Journals is proving as milestone of research publication. In view of whole spectrum of Knowledge, the research work of different streams may be considered as branches of big tree. Every branch is of great importance. Thus, we look after the complete spectrum as whole. Global Journals let play all the instruments simultaneously. We hope, affords of Global Journals will sincerely help to build the world in new shape.

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Advance Security warning system: Wireless technologies

Dr. Hameed Ullah Khan

***Abstract*- This Article Communicated An Approach For Determining The Precise Position Of The Person By Applying Smart Card Contactless And Wireless Communication Technologies. To Accomplish This Idea The Whole Scenario Is Divided Into Three Phases. In The First Phase Data Is Collected From The Area, In The Second Phase Data Activation Takes Place From The Area, Whereas In The Third Phase Data Is Processed To Obtain The Required Results For The Target(S). This Paper Provided Details On The Second Phase. The Concept Of Wireless Communication And Smart Card Is Taken Into Account For The Activation Purpose. It Is Faster In Gathering Data About The Exact Location Of The Target. It Is Also Worth Noticing That This Approach Is Efficient, Accurate And Requires Very Less Processing Time As It Is Very Specific In Activation And Yield Results For Quick Actions.**

Keywords: Information technology, security system, wireless communication

I. Introduction

This is a proven fact in the modern medical sciences that all human bodies develop from a combination of stem cells, which matures to form a fetus and so on into a neonate. This procedure is natural, (until and unless, under certain circumstances where it may not be able occur naturally) the medical sciences adopt methods, such as that of a test tube baby, but still its final stage of conception is same as the natural process. Similarly, the processes of death or decay by different means are also same. Besides this, it is an established fact that every human on earth lives under the rules of law and secondly has the right of self defense. Than the question arises, that under what situation/circumstances, this human decides to commit this nonreversible evil act which is called suicide? Every person condemns this act. As it is said very correctly, those who commit suicide are abnormal humans, as normal human cannot accomplish such an act. So the basic difference is between normal and abnormal humans. Normal humans can control their emotions which are link to the limbic cortex systems in his brain, where as abnormal are not able to control his emotion. Many reasons may push him to this last decision/choice, such as, economical conditions, stress & tension from society, religious radicals' pressures, weak personality, mentally ill, drug addicts, uneducated, deprived [1 - 7]. In the war against terrorism, the immediate short-term objective is to reduce the incidence of terrorism by using counter-force. To successfully do so, access, collection and timely interpretation of intelligence information is critical. Technology can be deployed to secure, control, and deny critical access and information, in order to reduce the capabilities to inflicting damages. The current trends in technological development point toward a combined use of several technologies such as biological technologies, robotics, information technology, and nanotechnology in the fight against global terrorism. Developments in information technology facilitate data collection, analysis, security, and integration; robotics can facilitate remote surveillance, the distancing of dangerous substances from human control, while biotechnologies can facilitate identification of biological hazards, provide forensic tools [8 - 10].

Similarly, there is a budding collection of research in the computer and information sciences domains that addresses new algorithms, techniques, models, and methods for engaging in the battlefield with insights on everything from sensor and laser technologies to complex information discovery models. Research on terrorism is housed in the legal domain, philosophical studies (especially in ethics and law), management (especially crisis management), health sciences, and engineering sciences. There is, however, a dart of cross-disciplinary research that involves meshing of two disciplines, e.g., computer science and public policy [11, 12].

In this paper three major fields are brought under discussions together. Prime work is based on wireless communication and its implementation from the application point of view. Secondly, the response or echo from the object by using laser beam and problem associated with obstacles are observed. Thirdly, the proposed approach is introduced which is different from the existing techniques and technologies. Also the proposed approach will provide advanced information irrespective of the hazards or obstacles.

II. Background Study

The information communicated in this section is based on wireless communication and application of laser technology:

A. Wireless Communication

In wireless communication, the data applications with affordable Quality of Service (QOS) over wireless networks are the demand in present and future generations. Scheduling at packet level in wireless networks is an important issue to be dealt. This motivates development of schedulers that can deliver the required QOS and being resource efficient. Wireless network, resource allocation

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schemes and scheduling policies are roles in providing service performances guarantees, such as throughput, delay, delay-jitter, fairness, and packet loss rate [13].
to improve/maximize system performance, e.g., throughput under various fairness and QOS constraints [15].

Wireless fair scheduling policies are discussed [16]. Several non work-conserving disciplines have been proposed: Jitter Earliest-Due-Date (Jitter-EDD), STOP-AND-GO Queuing (SGQ), Hierarchical Round Robin (HRR), and Rate-Controlled Station Priority (RCSP). Varieties of scheduling algorithms have been proposed for data services [17 - 21]. Scheduler's main task is to distribute the available bandwidth in a fair manner, among different simultaneous data flow scheduling policies of wire line networks are extended to wireless networks, where burst of errors in wireless channels are taken into account.

In this section, the detailed information is communicated regarding scheduler performance, function, operation of algorithm, classification, etc.

B. Networks and Performance

Schedulers are designed to multiplex a diverse set of packet flow and still be able to provide QOS [15]. In general, problem of scheduling in communication system are with possible exception of transmission errors (high error rate & burst errors, location-dependent and time-varying wireless link capacity, scare bandwidth, user mobility & power constraint of mobile hosts) [15]. Scheduling disciplines and associated performance problems have been widely studied in packet-switched networks [22].
Switch/router which buffers the traffic on the out port, the buffered packets are scheduled, or multiplexed for transmission on the shared common line. In the above case scheduler has updated information of all the parameters that effect scheduling decision, e.g., Buffer states, QOS parameters, which show quickness, react on any event. Thus, performance is essentially determined by packet arrival process, queuing discipline & packet transmission

Scheduling algorithms are studied in detail [14]. Various scheduling schemes are developed – A common objective is

time. Best-effort traffic is equally important to study performance parameters; throughput & fairness are special interest.
Wireless communication network consists of nodes that communicate with each other over a wireless channel. "In fracture wireless network", such as cellular networks, are widely prevalent, typically consist of wired infrastructure of controllers (base stations), with nodes connected over wireless link. "Infrastructure-less networks", such as ad-hoc networks, consists of purely wireless links.
Ad-hoc wireless networks allow speedy deployment, low cost & low maintenance, which lead towards applications such as sensor networks, personal area networks & military battle-field communications.
Scheduling is dealt in design of wireless network, at the link layer & relaying on data packets (routing) at network layer. Medium Access Control (MAC) is the process of scheduling the shared wireless channel between competing nodes.

C. Scheduler Components and Properties

In Wireless environments, the scheduling task becomes more difficult if channel conditions are taken in account, scheduling algorithms provide mechanisms for bandwidth allocation and multiplexing at packet level. Wireless scheduling algorithms, specifies algorithms consisting of five components [23, 24].
Algorithms can be used for any of the listed below mechanisms:
1) Error-free service model
2) Lead & Lag model
3) Compensation model
4) Slot Queues & pack Queues
5) Channel monitoring and perdition

Scheduling algorithms are important to provide guaranteed quality of service parameters such as delay, delay jitter, packet loss rate, or throughput.

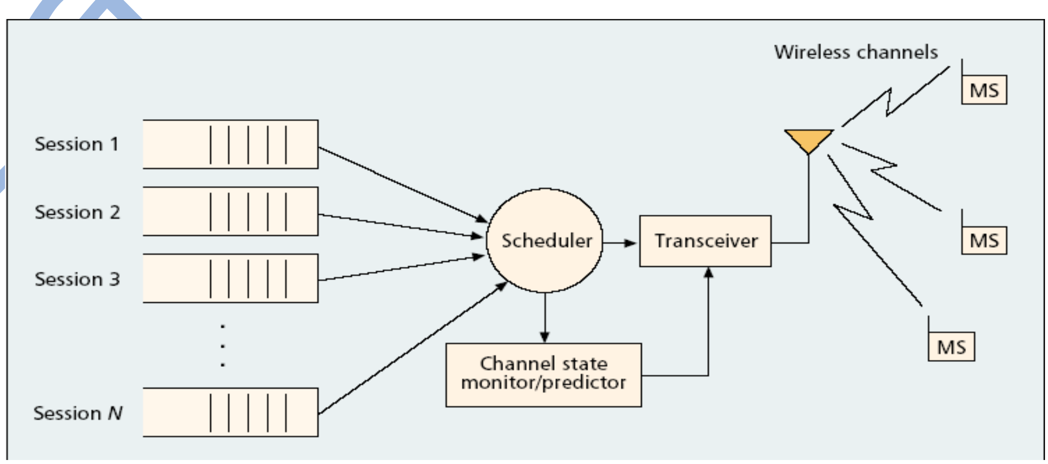

Figure 1: Typical Scheduler

In Figure 1, the scheduler operates across different sessions (connections or flows) in order to ensure that reserved throughputs are met. Main function of a scheduling algorithm is to select the session whose head-of-line (HOL) packet is to be transmitted next [22].

D. *Classification of Schedulers*

Schedulers can be classified as *work-conserving or non-work-conserving* [23, 24].
***Work-conserving* scheduler:** is never ideal if there is a packet awaiting transmission, e.g., Generalized Processor Sharing (GPS), packet-by-packet GPS also know as Weighted Fair Queuing (WFQ), Virtual Clock (VC), Weighted Round-Robin (WRR), Self-Clocked Fair Queuing (SCFQ) and Deficit Round-Robin (DRR) [22].
***Non-work-conserving* scheduler:** may be ideal even if there is a back logged packet in the system because it may be expecting another higher-priority packet to arrive, e.g., Hierarchical Round-Robin (HRR), Stop-and-Go Queuing (SGQ), and Jitter-Earliest-Due-Date (Jitter-EDD). Non-work-conserving schedulers generally have higher average packet delays than their counter part work-conserving.

E. *Scheduler performance*

Following are a few main scheduler performances:

***Time-stamped* scheduler:** is one that serves packets according to their timestamp values. Incoming packets are time stamped before being placed in their respective session queues. The HOL packets are sorted in increasing order of their timestamps, and the packet with the lowest timestamp value is selected for transmission. Time stamped schedulers can provide better QOS guarantees.

***Round-robin* schedulers:** do not use timestamps and can be more easily implemented.

***Sorted-priority* scheduler:** each session has a different priority level and packets are chosen for transmission according to their session priority, e.g., VC, WFQ, and Jitter-EDD.
***Frame-based* scheduler:** time is divided into frames of fixed or variable size. Each session reserves a portion of the frame for transmitting its packets.

F. *Cell-structured wireless networks*

In these networks, the service area is divided into cells, and each cell has a base station. Cell mobile hosts communicate via the base station, and base stations are connected via wire line networks.

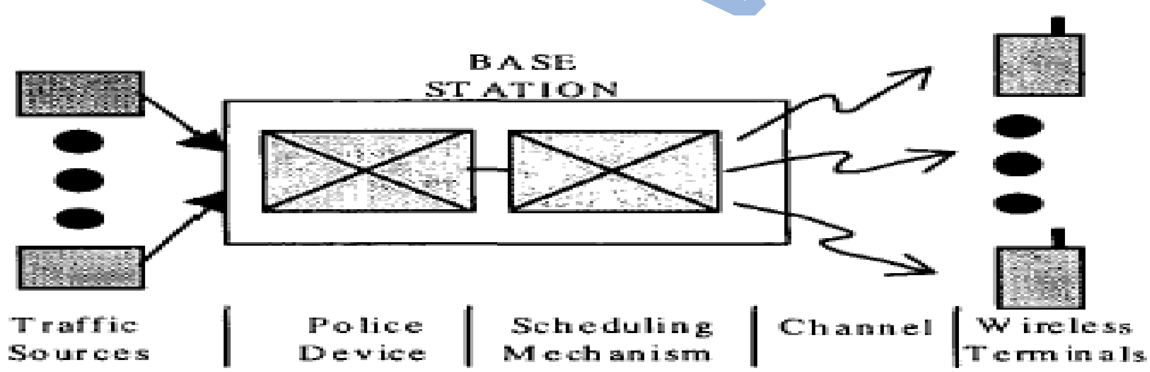

Figure 2: Base Station

In Figure 2, the base station is responsible for scheduling both downlink (from base station to mobile hosts) and uplink (from mobile hosts to base station) packet transmission between the mobile hosts and itself. The communication between a mobile host and a base station may consist of more then one traffic flow (or session). The wireless links between a base station and each of the mobile hosts are independent of each other.
Wireless links are subject to burst errors. A two-state Markov channel model is used for the state of wireless link, which is either of the two states: *good state (error-free)* or *bad state (error)* [23].

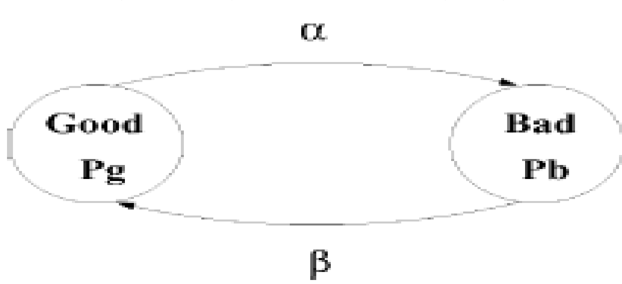

Figure 3: Good/Bad states.

In Figure 3, the transmissions between the two states occur randomly. In a *good* state (Pg = Good packet), the wireless link is assumed to be error-free. If a link is in *bad* state (Pb = Bad packet), packets transmitted on the link will be corrupted with very high probability.

G. *Application of Laser Technology*

The use of LASER Certainty for detection in every encounter, even if that reduces the possibility of advance warning. One must choose between these alternatives since they turn out to be mutually exclusive. A blend of both is possible if it accept that neither will be maximized [25].

In the real world, every laser encounter is different. The complicating factors are:

- The amount of sunlight (direct, reflected, or scattered) in the detector's field of view,
- The color of the human/vehicle in which the detector is installed,
- The distance from the laser gun,
- The location of the laser trap, and
- The space between the laser trap and the detector.

Sunlight interferes with laser reception. You can regard sunlight as the equivalent to background noise while trying to listen for weak sounds. Laser gun photons and the near-infrared components of sunlight are indistinguishable. Detrimental sunlight can be directly shining into the detector (the worst), or reflected and scattered by the world at large. The V1 has a specially tailored field of view to exclude "background glow" from areas that are unlikely to contain laser signals (obstacles, way high, way low, or off to the side).

The color of the human dress or vehicle can influence the amount of sunlight reflected from the hood into the detection optics. White, silver, and bright or metallic colors in general are 100 times more likely than to reflect competing light into the detector.

The pencil-thin beam of the laser gun expands with distance. The actual beam is only several feet wide at 500 feet, but "aiming wobble" introduced by the human holding the bulky gun make the detectable beam at least three times larger. Even without the wobble, it covers most of the place/roadway after a mile. This means it is actually much easier to find a fragment of the beam at greater distances from the laser trap. At short ranges, the beam may be so concentrated that a detector mounted away from the aim point on the human/vehicle may not be able to pick up enough stray energy to activate an alarm [26].

The detector needs a fragment of the direct laser beam, or at least a low angle reflection or glint of it. The beam is a straight line. If, for example, the laser is measuring human, the beam cuts across the obstacles, touching it only briefly. If the obstacle is even moderately tight, all laser energy that misses the target will soon be off the location and therefore out of play for the detector in any other person. While that setup may sound ideal as an enforcers' strategy, it also increases the probability of phony readings because of swiping error.

Obstacles ahead may block (bad), or reflect (good), the laser energy from a distant laser trap. The chances of blockage are considerably greater than a lucky reflection. Light obstacle is an advantage because there are other obstacles ahead serving as bait for the laser. A shot at them is a chance for detector to warn. Heavy obstacles, especially if dominated by any material, is bad because there can be nearly 100% line-of-sight blockage until its turn to be the target.

Now that we have the facts, let's put them into a defense system. If one's objective is never to miss a laser encounter, it is recommend for mounting the detector as close to the typical laser aiming points as possible. Here in the US, most laser operators aim at the front license plate since it is usually made with a special reflective treatment to aid night-time visibility (works great for laser visibility too, day or night). If there's no front plate, then a piece of bright metal can be the next most promising aiming point.

Advanced warnings (a warning in time to do some good) are more likely if one can avoid some problems caused by the low position. Advance warnings require the greatest possibility of receiving a weak fragment of the beam while it is being used on a good distance ahead. To maximize reception of weak signals, one must reduce blockage and interference.

Mounting high on the position the detector to "see" through the windows of most buildings/vehicles. Glass is a good thing for the detector to see; it can scatter reflections in direction. Sometimes it develops a glow, or a bloom, when the beam strikes, giving yet another chance for advanced warning.

Mounting high on effect of sunlight reflected from the hood, which increases the detector's ability to find weak beam fragments. Daytime warning sensitivity is completely dominated by how one manages the sunlight.

Mounting high behind reduces the laser signal somewhat, but it reduces the sunlight contamination by the same amount. The result is an unchanged laser-to-sunlight ratio from the scene beyond, but with reduced hood reflections owing to a higher vantage point—a net gain.

The downside of the move away from the most common aiming points, this means that in bright sunlight, you may miss the occasional direct hit from very close range—no detection at all! But that's possible with a low obstacle as well, just not as often. Finally, a warning too late is the same as no warning.

The proposed approach will pass on the information irrespective of the above hazards or obstacles mentioned above while facing in the use of laser approach [27 - 29].

III. Proposed Approach

Many countries, are using different approaches, such as scanning machines, introduced identity cards, walk through gates and passports with a fingerprint and facial (eye) biometric/facial-recognition and fingerprint-biometric technologies for their immigration control, etc., but it gives only the information, whereas advance actions have not been developed yet to stop the attacker. The attacker always assaults before the security personal reaches him or during the search. Another experience is also reported which is common among these attackers is that they don't leave any identity such as social security card, national identity card, etc, so it's not possible to find his nationality, address, etc. Let suppose, Figure 4 below represent's a crowd, where every small circle represents a single entity, as a human. The size of Figure 4 may be any it will not reduce the efficiency of algorithm, only the number of Main-Grids will increases which will cover the size of crowd.

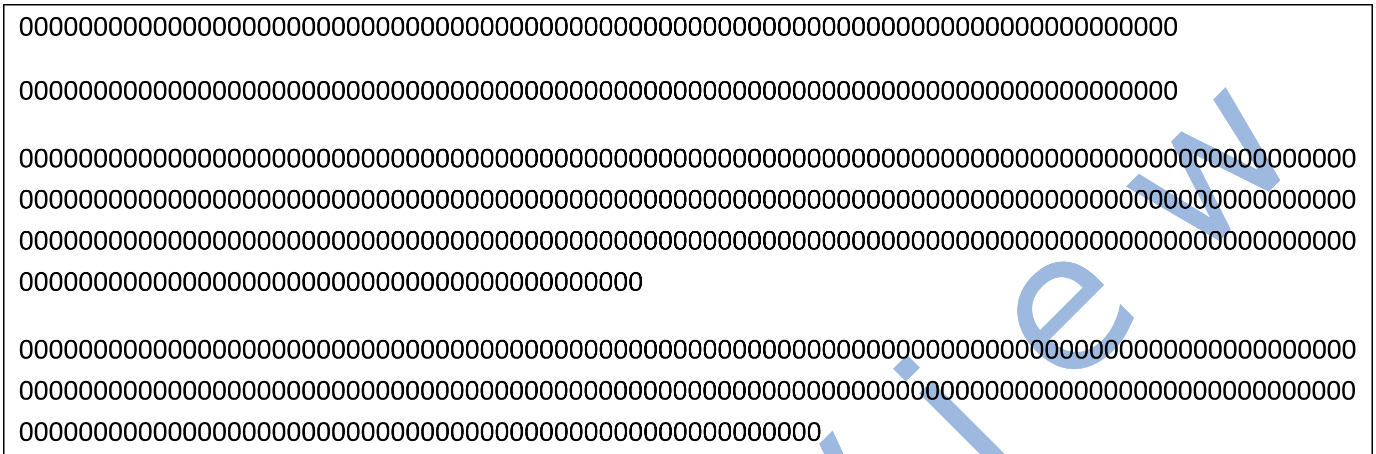

Figure 4: Crowded area

As the Main-Grids used on top of the crowded areas, through which each Mini-Grid will be activated and then the cells in-side will be activated, which is the source of information. Each Main-Grid composed of 10 Mini-Grids, starting from 00 and terminating at 09. In return each Mini-Grid comprising of 10 x10 cells. Each cell will be reporting for 100 people. Therefore, the complete one Mini-Grid will have the capacity of 100 x 100 people for reporting [29].

The proposed approach has to find the exact location of the attacker(s). For this purpose the algorithm is applied to collect information from the crowded area by using Mini-Grid, as shown in Table 1.

Table 1: Main Grid & Mini-Grids with Sub-sections

Main-Grid-0												
Mini-Grid-00			Column$_0$	C_1	C_2	C_3	C_4	C_5	C_6	C_7	C_8	C_9
	Sub-Section 0	Row$_0$	$G_{00}R_0C_0$									
	Sub-Section 1	R_1										
	Sub-Section 2	R_2										
	Sub-Section 3	R_3										
	Sub-Section 4	R_4										
	:	R_5										
	:	R_6										
	:	R_7										
	:	R_8										
	Sub-Section 9	R_9										

To achieve information, Table 1 is traced on top of Figure 4. The division of crowd will be reported as shown in Table 2.

Table 2: Combined representation

Main-Grid-0												
Mini-Grid-00			C_0	C_1	C_2	C_3	C_4	C_5	C_6	C_7	C_8	C_9
	Sub-Section 0	Ro	000000000 000000000 000000000 000000000 000000000 000000000 000000000 0000000 000000000	000 000 000 000 000 000 000 000 000	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00
	Sub-Section 1	R_1	000000000 000000000	000 000	00 00	00 00	00 00	00 00	00 00	00 00	00 00	00 00
	Sub-Section 2	R_2	000000000 000000000	000 000	00 00	00 00	00 00	00 00	00 00	00 00	00 00	00 00
	Sub-Section 3	R_3	000000000 000000000	000 000	00 00	00 00	00 00	00 00	00 00	00 00	00 00	00 00
	Sub-Section 4	R_4	000000000 000000000	000 000	00 00	00 00	00 00	00 00	00 00	00 00	00 00	00 00
	:	R_5	000000000 000000000	000 000	00 00	00 00	00 00	00 00	00 00	00 00	00 00	00 00
	:	R_6	000000000 000000000	000 000	00 00	00 00	00 00	00 00	00 00	00 00	00 00	00 00
	:	R_7	000000000 000000000	000 000	00 00	00 00	00 00	00 00	00 00	00 00	00 00	00 00
	:	R_8	000000000 000000000	000 000	00 00	00 00	00 00	00 00	00 00	00 00	00 00	00 00
	Sub-Section 9	R_9	000000000 000000000 000000000 000000000 000000000 000000000 000000000 0000000 000000000	000 000 000 000 000 000 000 000 000	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00	00 00 00 00 00 00 00 00 00

As explained earlier, each cell has the capacity of 10x10 people to cover. For example, notice at position Row-9 and Column-0 (R_9C_0) it records 100 people. Therefore the total capacities of one Main-Grid for only one Mini-Grid will have 100 x 100 people. In case if an area is large enough and can not be covered by one Mini-Grid, then another Mini-Grid will be applied to cover the area. And if one Main-Grid is not sufficient, then more Main-Grids are applied with the same structure but different Main-Grid addresses.

A. Mobile Technology and Smart Card

Mobile phones are the backbone of mobile communications and have experienced explosive growth. Currently, the third-generation mobile phones (3G mobiles) that aims to offer high-speed 2M/second communication using a greater transmission efficiency in the high frequency 2GHz band and more. This will allow multi-media communication (such as animated images), a typical example of which is viewing TV on a mobile phone.

Imminent is the practical application of new technology called "3.5G" which realizes further high-speed data communication by developing the technology base of the 3G mobile. This not only realizes maximum 12Mbps transmission speed but was developed with fixed-rate communications in mind, so together with the high transmission speed of maximum 12M bits/second, a cheaper and more comfortable mobile environment is about to be realized.

There are many different names for the smart card, for example IC-card, microprocessor card, electronic card, etc. Nowadays Smart card contains semiconductor device (chip) and a data link for data communication between the smart card and databases. Based on needs and use smart card is of two types as explained below.

B. Schematic of Contact Smart Card

Contact smart cards have a contact area, comprising several gold-plated contact pads, that is about 1 cm square. When inserted into a reader, the chip makes contact with electrical connectors that can read information from the chip and write information back. Figure 5 shows a common version of a contact smart card. A cavity is produced in the card body and into this cavity a chip module is bonded with an adhesive. The chip which is connected to the outer side contacts through thin gold wires. The contacts are defined by international standards in their number, size and position, hence the function of each smart card reader is guaranteed over the world [30].

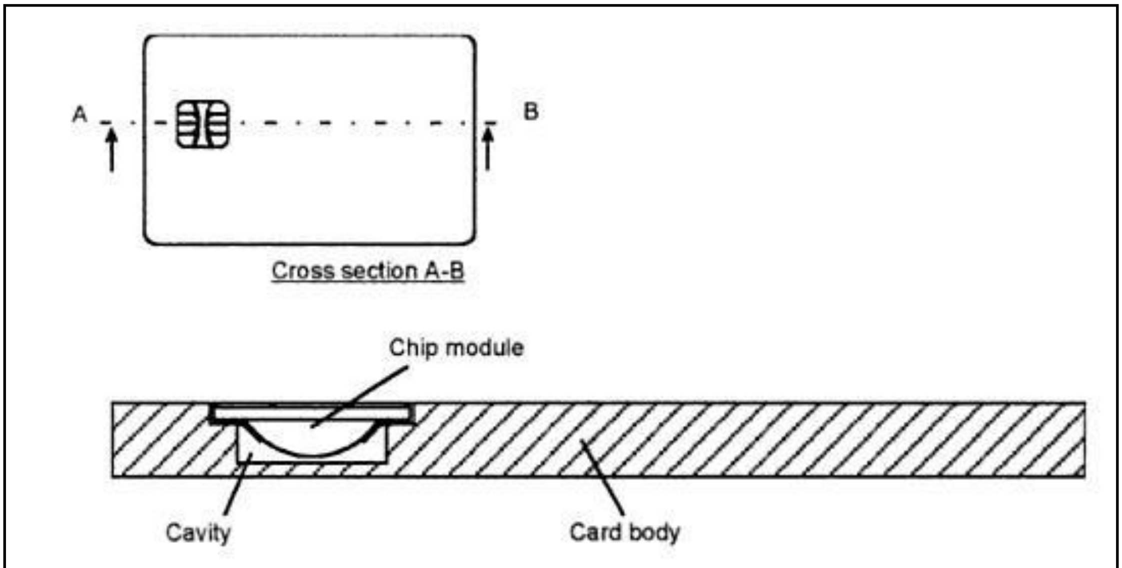

Figure 5: Schematic of Contact smart card

C. *Schematic of Contactless Smart Card*

A second type is the contactless smart card, in which the chip communicates with the card reader through RFID (Radio Frequency Identifier) induction technology. These cards require built-in antenna to complete transaction. They are often used when transactions must be processed quickly, such as on mass transit systems. Figure 6 shows a common version of wireless contact smart card. In these cards a subscriber identity module (SIM) for the identification of the user is made in the network. Typical applications are tickets for sports events, public transport systems and security checks [31].

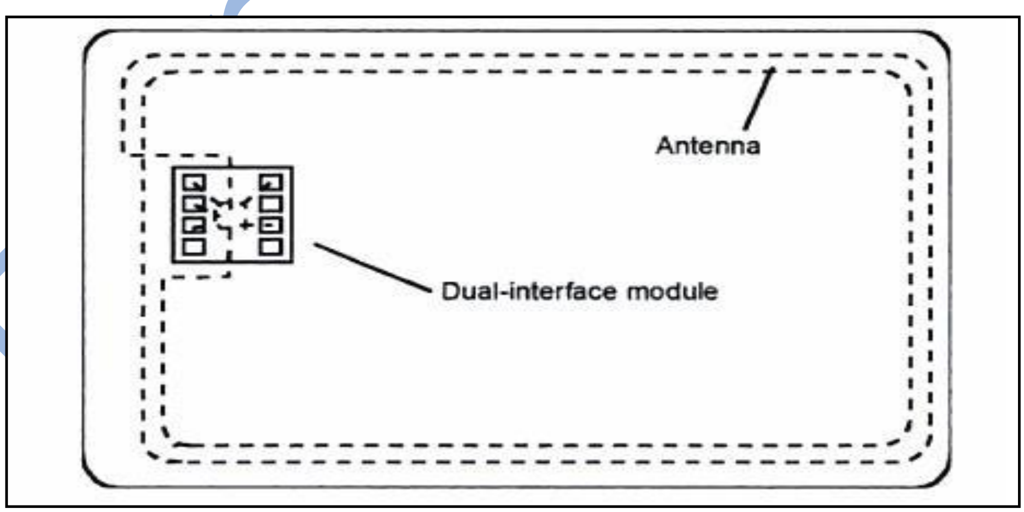

Figure 6: Schematic of smart card without contact

D. Proposed Approach

The proposed approach for activation information depending on the contactless smart card, as shown in Figure.7

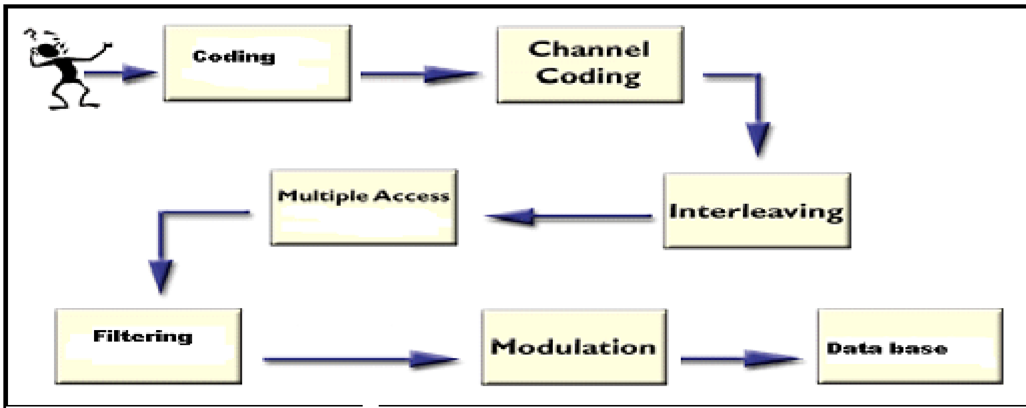

Figure 7: The proposed approach structure

Coding; is performed in a digital communication and therefore it must be converted to a digital bit stream and then uses the redundancy in the signal and achieve a bit rate. Channel coding; once the signal has been coded into a digital bit stream, extra bits are added to the bit stream so that then it can be recognized and correct errors which could have occurred during communication. Interleaving; is the processes of rearranging the bits, it also allows the error correction algorithms to correct more of the errors that could have occurred during communication. By interleaving the code, there is less possibility that a code can be lost. Multiple accesses; allow many codes at the same time. The transmission of the signal is continuous but the data is transmitted in series. The assembly operation takes the final encoded data before the database. Filters; are used to remove noise, which also give early indications for possible existence of targeted data. Modulation; Modulation changes the '1' and '0's in a digital representation to another

E. Smart Card Activation

Smart card is based on the mobile technology approach as shown in Figure 8. When the signal is sent; to the Main-Grids mainly people, the response monitored in shape of echo signal received then it is 100% clear place. Because on many occasions, the attacker doesn't leave any traces of information leading to his identity. They do not leave any record at all to give evidence/details of that person.

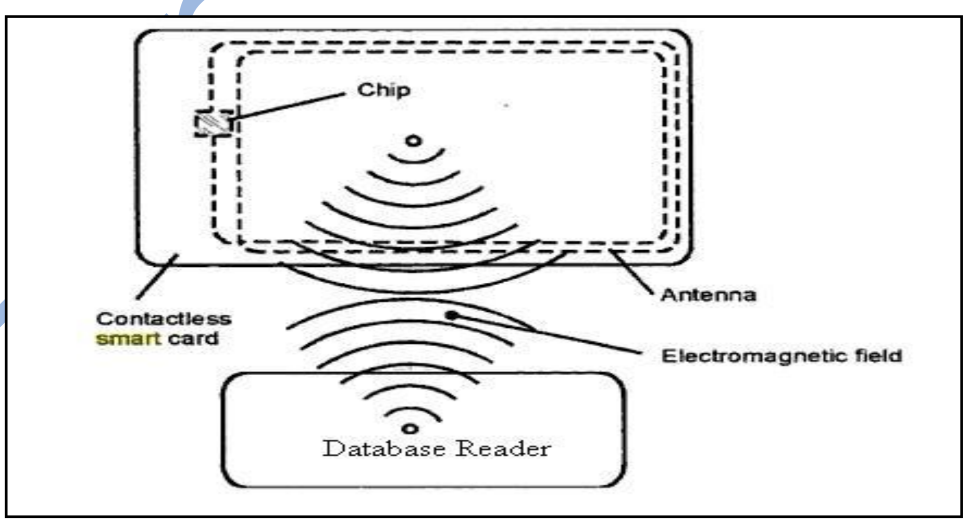

Figure 8: The proposed smart card activation paradigm

]On the other hand the damage is not stopped. Moreover, he got out of the situation and his mission was accomplished. Our concern is to reverse the situation by stopping him from both destruction and to catch the attacker alive; the later case is not yet possible. But if the signal received is negative from one or more locations than these are the expected places of such individuals.

F. Controlling the attacker

The proposed approach gives the concept of cage/bubble to be used at the time of search and to curb the attacker. It is reported that attacker normally blows himself up at the spot while security personnel are busy in searching his body. To control such situations it is suggested to use the cage to cover the attacker first. The cage is designed in two types as shown in Figure 9.

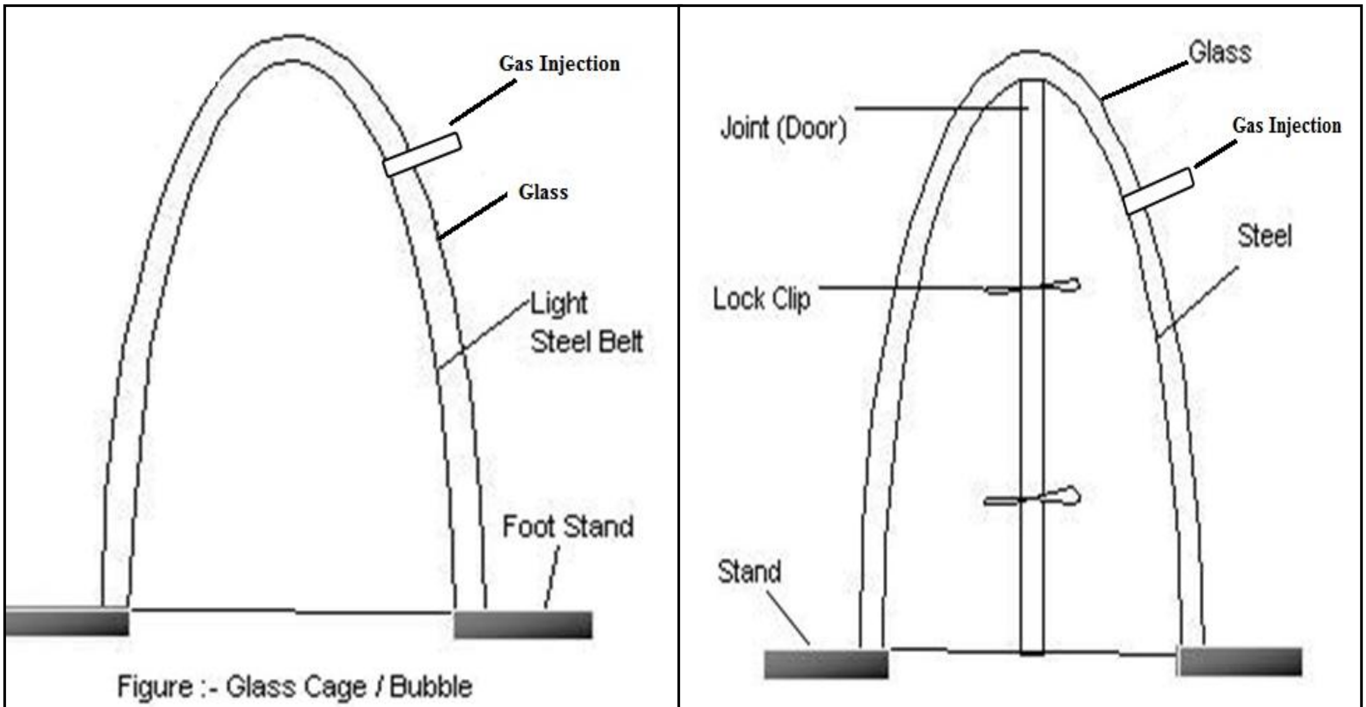

Figure :- Glass Cage / Bubble

Design (a)

Design (b)

Figure 9: The cage/babble design (a) & (b)

The inner side in both types are made up of light weight, fire protected steel and the outer side is made up of bullet, chemical, heat and shock proof glass. So the search carried out inside the glass cage while the security person is standing out side on the foot stands out side the cage to search the attacker by asking him to take off his shirt because jacket/explosives are always placed on the abdominal part of the body and occasionally a small fire arm that can be seen easily from the out -side as the cage is transparent. Normally the attacker attitude is very non-cooperative, therefore a gas tube is places on the top of the cage so if he refuses to obey the orders then mild gas (injected as an inhaler to give anaesthesia) will be injected through the inlet in the cage to make the attacker unconscious for a short period and it will be possible to curtain him without causing any damage [32].

The design of cage is of two types: one is in single piece and other in two pieces. Two pieces means that it opens when one stretches his arms in outer direction and if you wanted to close then press the arms in inner direction. There are hinges so it is half open and when engulf some one it is closed.

IV. Conclusions

The use of cage seems to be impossible. In case if the attacker is not stopping then the security can injure him by shooting him on his legs and can then place a cage on top of him. So if he is for sure an attacker, than he is incapacitated while inside the cage.

The foremost approach is to look into the problems of such people, whether it's their economical problems, it's a stress based, religious based or else, but it must be resolved rather than using force to eliminate. If one is finished then many other emerge. Deaths of such individuals only serve to inspire others of like mind and also as rallying calls for others. This is a very wrong strategy by using force; matters must be resolved at grass roots level by dialogue and try to use psychology rather then force on human. This has also been noticed that during the attack, he is mentally ready to blow up his life/body and give maximum damage. The situation is to handle a surprise which is almost impossible to know before hand what is in his mind. Therefore the use of modern approaches and technology must come in use to reduce damage as surprise on the attacker is reverted so he will be curtailing before hand. As some one said very correctly, do it to them before they do it to us. That's why the quick action along with prior awareness to handle the prevailing conditions and warning, they must combine together to handle such sudden situation. Again some one said wisely, a warning late is the same as no warning.

V. Future Directions

Scheduling in Wireless Networks is applied in the latest technologies of wireless communication, such as Bluetooth, Infra Red, etc. It can be used and enhanced in other related areas of communication. The proposed approach will apply in many other areas for information gathering even helping police departments to reduce the crime rate.

VI. Acknowledge

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Achieving Network Layer Connectivity in Mobile Ad Hoc Networks

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***Abstract*—Mobile Ad Hoc Networks (MANETs) is an area of networking which has been the focus of intense research in the past years. Due to their differences from traditional wire line networks, MANETs require a completely different set of protocols to cope with their decentralized nature. As such both evolution and innovation is required in many sectors. One such sector is the network layer which encompasses numerous important functions. This paper focuses on providing a comprehensive guide on achieving node connectivity at this layer. This includes selecting a proper routing protocol, as well as an autoconfiguration algorithm. These are assumed to operate around an IP protocol, more specifically IPv6. Finally we will discuss possibilities for ensuring QoS in Ad Hoc networks.**

Keywords: ad hoc routing, QoS, auto configuration

I. Introduction

Mobile Ad Hoc Networks are considered one of the most promising areas of networking. An Ad Hoc network consists of mobile nodes, which may vary in size & capabilities which communicate to create a network without pre-existing infrastructure. Thus a MANET can be formed dynamically without any pre-existing infrastructure, reducing both deployment time and costs and increasing flexibility. Unfortunately these advantages provide us with a set of problems. The majority of current network protocols have been developed to operate in strictly defined, mostly static environment, so using them in an ad hoc environment is the very least problematic. Thus a new protocol stack should be defined, using mostly newly developed protocols that can answer the challenges met in ad hoc networks. To define this protocol stack it is imperative that we develop a framework upon which the evaluation of such protocols can be accomplished. The network layer is responsible for converting the facilities of the lower layer into services that the upper layers can use. It is responsible for a host of important tasks such as routing and addressing and configuring nodes. The nature of Ad Hoc nature makes it impossible to use current network layer protocols. Thus a host of new ones have been proposed to achieve connectivity at this layer. This paper examines Ad Hoc routing protocols as well as address autoconfiguration algorithms. The former are protocols specifically developed to forward packets in multi-hop networks & the later aim to allocate each node in a MANET a unique IP address. Then we attempt to use these mechanisms to provide QoS mechanisms at the network layer. QoS is a required for a number of applications particularly real-time and critical ones, which are dominant in several areas of possible MANET use, such as military or aviation applications.

Mobile Ad Hoc networks are very different from wire line networks. In the later everything predetermined, that is the network topology is already know as well as its infrastructure and the equipment used. This allows for network administrator and architects to carefully plan its deployment to meet their requirement. Unfortunately Ad Hoc Networks are very different in that there is no knowledge about any of the abovementioned parameters. So there is no real information about the physical or logical connectivity of other nodes, neither about the services provided by each. This comes in stark contrast with traditional networks where most information is preset and those that aren't can be discovered with a simple service discovery protocol.

The rest of this paper is structured as follows: In Section II we will an overview of auto-networking technologies for MANETs. In Section III we will analyze Ad Hoc routing. Section IV will investigate the application of Quality Of Service mechanisms in Ad Hoc Networks. Finally Section V combines the above elements and provides the groundwork for future work.

II. AUTO CONFIGURATION TECHNOLOGIES FOR MANETS

One of the most important characteristics of Ad Hoc networks is their spontaneous creation. For this to be achieved a mechanism must be invented that is able to organize the network and manage resources (like IP address) and configuration parameters (like the maximum transmission unit – MTU). In most applications this is impossible to do manually. Configuring an Ad Hoc network at the network layer involves one fundamental task: Unicast Address Allocation.

Unicast Address Allocation is the first and absolutely essential goal of the presented auto-networking technologies. Without a unique network layer address unicast communication is impossible. Obviously a stateful method, such as DHCP cannot be used, because it is not possible to guarantee access to a DHCP server for each node and since introducing such an centralized component weakens one of the fundamental MANET advantages, namely distributed operation.

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The newest version of the internet network layer protocol IPv6 includes algorithms for both stateful and stateless each node, the verification of the uniqueness of this address through a Duplicate Address Detection process and finally the construction of a site-local address through the acquisition of a Router Advertisement message.

This algorithm while useful is inadequate for use in Mobile Ad Hoc Networks for several reasons. First of all it requires the presence of router on a link to configure anything but link-local addresses, but provides no means for auto configuring routers. In Ad Hoc networks all nodes play the role of a router thus it is practically impossible to use this algorithm. Nevertheless it has served as an inspiration for other mechanisms, some of which are described below.

The issue of node autoconfiguration (and in particular address allocation) has been the focus of significant research. Over the past few years numerous solutions have been proposed. These solutions can be subdivided into three categories:

A. *Conflict Detection Allocation*

Conflict Detection Allocation algorithms present the most straightforward solution to the problem of unicast address allocation. They adopt a method of trial and error to assign each node a valid address. The process is quite simple. The new node selects a random tentative address, then broadcasts a message to the whole network asking if that address is unique. If no response is received after a finite address autoconfiguration. This algorithm involves three steps: The assignment of a tentative link local address to number of retries the address is considered unique and assigned to an interface. If an answer is received then the selected tentative address is already occupied and the node must select a new one and repeat the process.

B. *Conflict Free Allocation*

Conflict Free Allocation algorithms assign each new node an address that is already known to be unique. This is accomplished by using disjoint address pools for each node. Thus there can be no conflicts among the allocated addresses. Obviously to accomplish this each node must keep some sort of state information for each address.

C . *Best Effort Allocation*

Best Effort Allocation algorithms attempt to assign a new node an unused – to the best of their knowledge – address, but still use conflict detection methods to ensure that this address is indeed unique. Each node keeps a state for each address, but because he cannot assume to always have up-to-date information regarding the entire network cannot be sure that the information upon which it bases its address allocation is valid.

Following is a table describing the most important characteristics of each algorithm:

	Conflict detection	**Conflict free**	**Best effort**
Network Organization	Flat/ Hierarchical	Flat	Flat/ Hierarchical
Overhead	High	Small	High
Network Settling	Time	High	-
Node Join Time	High	Small	High
Address Reclamation	Not needed	Needed	Needed
Node Depart Time	-	Medium	Medium
Distributed	Yes	Yes	Yes
Complexity	Small	Medium	High
Evenness	Even	Uneven	Even
Scalability	Small	Medium	Small

In short we can say that best effort allocation algorithms tend to be the least useful, that is because the actually combine the worst of both worlds. To elaborate a little on this:

There are two important setbacks for Conflict Detection allocation. Firstly it broadcasts information on the network and it does it quite often, resulting in rather large overhead and secondly there is considerable delay until an address is assigned to an interface due to the timeouts involved. Best effort allocation has these disadvantages. Conflict Free allocation on the other hand has neither but is usually quite complex to implement and requires that an address state table is kept thus consuming memory which is not abundant in mobile nodes. Best effort allocation also maintains state tables, which is an additional problem. In general we can say that best effort allocation can be successfully used only with proactive routing protocols so as to take advantage of their periodic signals to update it's state tables.

To conclude we can say that both Conflict Detection and Conflict Free algorithms have their advantages. Conflict Detection Algorithms tend to be less scalable than Conflict Free ones, though the later cannot provide really large scalability either. For simple networks consisting of a few nodes a conflict detection algorithm like the one proposed in [6] would be ideal. For more demanding applications, complex solutions must be devised, possibly combining advantages from several categories.

III. ROUTING PROTOCOLS FOR AD HO NETWORKS

A routing protocol must meet various requirements for its proper use in mobile ad hoc networks. Such requirements are low network and memory utilization, scalability, the ability to cope with increased node mobility, loop freedom, minimal routing overhead, Quality of Service capabilities, security and bandwidth efficiency.

Routing for MANETs has received the largest research focus in the past years. These efforts have yielded considerable results in the form of numerous protocols. These protocols can be classified into four categories: On-demand, Table-driven, Cluster-based and hybrid. Each of these categories follows a different approach and as such has its own different ups and downs. A short description of each category follows:

A. On Demand Protocols

On Demand protocols discover paths to a destination only when requested. Their function is compromised of two tasks. The first, route discovery involves finding valid routes to a destination. This is accomplished by broadcasting a Route Request (RREQ) packet on the network. This packet propagates through network until it reaches the destination node, which then retraces the route and replies with a Route Reply (RREP) packet. (Note that the route inversion is only possible when the links are symmetric). Since this is not always the case the node transmitting the RREP packet may also have to perform route discovery. When the node initiating route discovery receives a RREP packet it has at least one valid route to the destination node.

The second task that on-demand routing protocols must handle is route maintenance. This involves discovering and patching up problems with already discovered routes. This is handled through Route Error (RERR) packets that are transmitted when a node detects a broken link. Nodes receiving this packet stop forwarding packets using routes that use this link.

On-demand protocols have several advantages, the most important being low overhead, since routes are only discovered when requested. In addition since no routing tables are maintained they require relatively little memory to operate. On the downside they introduce a considerable delay from the request of a route until it's discovery. Examples of on demand protocols are the Ad hoc On Demand Distance Vector (AODV) and the Dynamic Source Routing (DSR).

AODV is the most sophisticated protocol for MANETs so far and has been at the epicentre of most research. AODV follows the on-demand protocol format described above. In order to avoid the infinite looping of packets of the "Bellman-Ford" algorithm, AODV uses sequence numbers to stamp routes from an originate to a destination node. AODV is also capable to manage security considerations and it has multicast and other abilities through the various existing extensions.

B. Table Driven Protocols

Table driven protocols maintain tables in which they attempt to have at least one valid route to each node in the network. This is accomplished by the periodic broadcast of messages. With these messages a node declares its presence and availability to its neighbours. When the network topology changes, nodes update their tables by transmitting update packets. These tables can also contain other useful information, such as a list of all the transmitting nodes neighbours or the nodes current routing table. The major strength of proactive protocols is that there is no delay until the route request is served. Their weakness is that they produce high overhead due to the continuous packet transmissions. An example of table-driven protocols is TBRPF (Topology dissemination Based on Reverse Path Forwarding).

C. Cluster based Protocols

Cluster based protocols are based on the concept of grouping nodes together depending various topology parameters. These protocols usually elect a cluster head node, which is responsible for the communication with other clusters. The connection between the different clusters can be achieved through intermediate nodes, known as gateways, which belong to many clusters at the same time. The advantages and disadvantages of these protocols may vary depending on the use of the ad hoc network. The most serious drawback is that they introduce a form a centralized structure which is difficult to maintain due to node mobility. On the upside routing overhead is significantly limited. An example of these protocols is the Cluster Based Routing Protocol (CBRP).

D. Hybrid Protocols

Hybrid protocols combine various characteristics of all the above categories. Depending on the protocol, we have on demand protocols with enhanced use of procedures of table driven protocols and the opposite. Many protocols also use clustering concepts depending on the application for which the mobile ad hoc network is intended. An example of these protocols is the Zone Routing Protocol (ZRP).

IV. QOS MECHANISMS IN AD HOC NETWORKS

The mobility and dynamic topology of the nodes in a MANET make network management a really challenging. This is because the level of the offered "quality" in an established connection varies depended of a variety of external conditions. So the intention is the definition of a Quality of Service (QoS) model which will operate with the minimum resources and will adapt troublelesly in dynamic environments.

QoS is the mechanism which is responsible for the management of traffic in such a way that it can meet the demands of each application which wants to use the network each time without wasting the already scanty in MANETs resources.

When we refer to the availability of QoS we mean a set of quantitative metrics which define it. These are the available bandwidth, the packet loss rate, delay, packet jitter, hop count, path reliability.

The use of QoS is essential in applications which are sensitive to the time of their transmission, such as real time applications. People will be using MANETs to connect each other via very common devices (PDAs, laptops, mobile phones etc.) from almost anywhere and use services such as video on demand, videoconference, and internet telephony.

Some additional difficulties for providing QoS in MANETs arise from their decentralized nature, their limited - due to the wireless links - bandwidth, the case of overload, the signal attenuation, noise, external elements, limited resources, power management, end to end protocols and demands of the applications.

Up to today most research on providing QoS for MANETs is the evolution of the two main architectures for wired networks, Integrated Services and Differentiated Services. The later dissever each flow of the traffic and treat each independently according to its demands, while the in former all the flow is been treated using a single method.

QoS metrics should be taken into account when designing a routing protocol. Usually these are either the minimum bandwidth or the maximum delay, as well as the method for path calculation, the way by which the QoS will be forwarded to the other nodes and remain stable and dissever priorities. All these ought to dynamically adjusted with each topological change of the network.

CEDAR (Core-Extraction Distributed ad hoc Routing Algorithm) is an algorithm which provides routing with quality of service in MANETs. To establish a connection the algorithm divides the network into smaller subnets in which the core extraction mechanism chooses an appropriate node to be responsible for route computation. The core nodes are then informed about the condition of surrounding and their bandwidth availability. The next step is the establishment of a connection between the source and destination nodes, considering the information provided by the core nodes. The main advantage of the algorithm is its simple routing structure, as well as the fact that it's cluster based architecture assigns most of the work to the core nodes. This architecture proves to be the algorithms main setbacks as these nodes can become overwhelmed in scenarios with high node mobility or a large number of nodes.

Research on the two aforementioned architecture had yielded a number of mechanisms for providing QoS, the most important of them being the ReSerVation Protocol, DiffServ, Multi Protocol Label Switching, Subnet Bandwidth Management.

RSVP is a very promising algorithm. It differentiates each flow from the traffic stream. A session defines the destination address, destination port and a protocol identifier. The messages needed for the propagation of the QoS metrics are transmitted to the same direction as the media flow. It supports both multicast and unicast flows, which are reserved in one direction only. It is a soft state, receiver oriented protocol, which allows transparent flow through non-RSVP routers and switches. RSVP does not control directly the behaviour of the network devices.

Another way to establish QoS conditions in a network is the through signalling. INSIGNIA is the most prominent signalling protocol. It is quite effective since it accomplishes not to use many acknowledgment packets thus not imposing a significant amount of additional overhead. It also includes a feedback mechanism, which decreases the error probability.

Finally the use of IPv6 as the default network protocol provides as with some built-in QoS capabilities, through an option in the hop by hop extension header (QoS Object Option).

V. Conclusion

In this paper we described numerous technologies that attempt to answer the most important challenges met in the network layer in Mobile Ad Hoc Networks. These technologies can be combined in various ways to achieve the desired result, which is a reliable network layer protocol under the IPv6 umbrella.

Future work includes the realization of this combination and it's incorporation in a complete protocol stack.

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Impact of the IT Revolution: A Case Study of King Saud University

Yousef Al-Ohali

***Abstract*: The current era is considered to be an age of information. The technologies to convey, retrieve and manipulate the information have gain tremendous attention especially in business-oriented environments during the present decade and academia across the globe are no exceptions. The academia across the worlds seems ready to be part of the current Information Technology (IT) revolution that is changing the world quickly and has eased up the access to information. In order to provide the incomparable advantages of e-services and related IT technologies, the university's visionary leadership decided to heavily invest on the focused areas of IT revolutions.King Saud University (KSU) has delivered a lot in past few years as the result of extensive funding that have been spent on the establishment and enhancement of the e-services environment and infrastructure development. Keeping forth the growth of the university the webometrics 2010 has kept KSU at 199th position among the best universities in the world. This paper focuses on the past, present and future of the KSU in terms of the establishment and enhancement of the electronic environment.**

Keywords: KSU, e-services, e-infrastructure

I. Introduction

King Saud University (KSU) is the largest yet and eldest university of the kingdom established in 1957. The university has a huge desire to remain committed with its mission and values that are focused around the best technological education and prosperous working environment for the faculty and staff within the Islamic culture and traditions. The idea of establishing the first university in the Kingdom of Saudi Arabia came as a natural response to the educational revival the kingdom enjoyed since its foundation. With the reign of King Abdulaziz, the kingdom's government sought to propagate education throughout the country. His efforts were crowned when King Fahd bin Abdulaziz assumed the office of the first Ministry of Education. Right after the first session of the Council of Ministers, he gave the following statement: "We will shortly establish the Saudi University; this is a foregone conclusion. This university will be one of the most prominent houses of culture and sciences, and will be worthy of our country where the light of Islamic faith and civilization has emanated."On another occasion, late King Fahd said: "I am seriously interested before anything else in supporting higher and vocational education in the country to have history repeat itself and add a new glory to our glorious past. Therefore, the establishment of the Saudi university with all its colleges, institutes and laboratories built according to high standards, is my immediate concern." Only three years after the establishment of the Ministry of Education he announced in a statement to the press the following: "The Ministry of Education is seriously considering the creation of the Saudi University." Approximately one year later, the first Saudi university was founded according to the dictates of the Royal Decree No. 17 dated 21/3/1377 H and read as follows: "With the help of Allah, We Saud bin Abdulaziz, King of the Kingdom of Saudi Arabia, having the desire for the dissemination and promotion of knowledge in Our Kingdom, for widening the base of scientific and literary study, and for keeping abreast with other nations in arts and sciences and for contributing with them to discovery and invention, and to revive Islamic civilization and articulate its benefits and glories, along with its ambition to nurture the young virtuously to guarantee them healthy minds and ethics. The university was named as King Saud University, established in the capital city of Riyadh. The university, since its birth in 1957, has become the first choice for higher education for the local and regional students. Currently the university is proudly considered at 199th [1] position among the best universities in the world, and pursuing for even better position. This top position has become as a result of the top management commitment, which has a very clear vision and leadership to help this university grow.

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II. Vision and Values

The university has a very clear vision to excel academically while following the Islamic cultural values, the university is committed to [2]:

- Quality and excellence: is measured according to challenging criteria, honoring high ambitions and the pursuit of distinctiveness through commitment to the highest intellectual standards in teaching, learning and innovation.

- Leadership and Team Work: The university remains committed to promoting individual and institutional leadership roles that drive social development upholding professionalism, responsibility, and innovation. Also it aspires to maintain team work all along.

- Freedom of inquiry: Rigorous and honest intellectual exploration is fundamental to the university's academic traditions, and it is reflected in all the dimensions of scholarly activities.

- Fairness and integrity: the university is proud to abide by the principles of social justice, equal opportunity and cultural diversity, consequently holding the members of the community to the highest standards of honesty, respect, professionalism and Social and professional ethics.

• Transparency and accountability: KSU remain committed to expose the thinking and ideas for society and scholars to judge the university's contributions to global knowledge, and KSU hold accountable everybody in our community for respecting and upholding the values in all forms of their scholarly activities, in the best way.

• Lifelong learning: KSU is committed to lifelong learning inside and outside the KSU community, enhancing continued intellectual growth and welfare of the society, in the best possible way.

III. Top Management Commitment & Support

Top management of the university has been very focused since the establishment of the university. The university has blossomed a lot in the era of the current rector of the university: Dr. Abdullah Al-Othman. The vision, commitment, loyalty, sincerity and hardworking of the rector and his devoted team has made the university to achieve and strive for the technical excellence. The university management has decided a number of goals to achieve and for that purpose keen efforts are being made to achieve them. The goals include but are not limited to

- Identifying and determining the key issues affecting the university mission
- Improving the internal efficiency of the university and quality of education
- Providing an environment attractive to distinguished faculty
- Satisfy assessment and academic accreditation criteria at the national and international levels
- Provide standard research requirements and resources and encouraging researchers to distinctive international fame
- Provide appropriate educational environment suitable for implementing modern technologies and trends in university education and perception.
- Enrich and enforce partnership between KSU and various community sectors
- Strengthen, Enrich and propagate the culture of strategic planning [3] within colleges and university administrations

The technical excellence has been achieved by execution of a number of programs including but not limited to:

- Distinguished Scientists fellowship program
- ISI KSU awareness
- King Abdullah Institute for Nanotechnology (KAIN)
- Riyadh Techno Valley

All these programs are contributing to implement the strategic plan [3] of the university and are focused to provide distinguished research environment.

IV. Past, Present and Future

It looks highly imperative to discuss the impact of the decisions that has been made an the management level and how they have effected the standing the university nationally and internationally. Regarding the up gradation of the e-facilities across the campus. It is also important to note that five years ago KSU was using legacy ICT system which has been replaced with the state of the art system currently and still posses more room for further enhancements in the future to become even better and more acceptable to the stakeholders of the university. In 2005, there was no concept of providing the e-services that certainly had a lot of anomalies and duplication of data across the university. in 2005 the basic facilities like e-mail were just limited to the faculty only. In recent time, the KSU management has broadly focused to make the changes visible to the world, of course, in the presence of the brighter vision the dynamic leadership of the honorable rector, new programs are being initiated and events are happening dynamically. In order to make the access to information more convenient and reliable it is important that the end users and the visitors are provided with the access to a web portal where they can find the information that is more relevant to them in terms of their needs. For this purpose the KSU web portal has been developed that in order to establish a web presence that reflects the image and reputation of King Saud University and its educational and research activities, the university has initiated the KSU Portal project. This online portal, enhanced with the state-of-the-art technologies available today, provides all with an enriching experience.

KSU Portal is designed to be a knowledge gateway for its faculty members, students, and visitors from all over the world. Its content is the result of contributions made by its different departments, deanships, colleges and staff members. Each college in the university has its own bi-lingual website that contains information about the college, departments, activities, and educational material. Again, more than 4600 Faculty members update their websites with their scholarly publications, articles, and course-related materials. Their websites have become an outlet for sharing their contributions and providing their students with a valuable educational experience.

The KSU portal has already led to several advances. For instance, KSU is the first in the Middle East region to have its campus available in 3D image on Google Earth environment, following the lead of highly ranked universities. The portal also contains an interactive map that can help in identifying the location of each college and respective buildings with respect to their physical location. The portal contains the sub sections about the students, who can avail the e-services provided by this portal, including: the salary check, record updation, and login to the system. Almost all forms that are required by the faculty and students have also been made available on the portal to facilitate the masses to provide maximum benefits to the concerned people and departments of the university. Despite many other features of the portal the weekly news magazine of the university is also available on the portal for consistent and constant viewing of the users. The portal also maintains the archive of previous magazines. The KSU also has taken the lead in broadcasting some of its activities live over the Internet. A case in point is the broadcasting of the first Dental health care conference; such broadcasts not only provide people an easy access to knowledge, but also provide an international interaction among peers.

KSU plans to continue this wonderful advancement by enhancing different parts of the Portal. The Knowledge Center part of the portal is designed to be always updated with different materials intended for different audiences ranging from students and non-academic people to leading researchers in different fields.

The portal has been implemented in Arabic and English to provide more efficient local and global access.

V. Advancement Comparisons

King Saud University (KSU), in response to the initiatives taken by the management has flourished dynamically in recent past, and expected to blossom hugely in the near future. Different factors that help in implementing the IT services, including network bandwidth, wireless coverage, network nodes and e-services can be compared to see the advancements that have been made as yet and that are expected to be made in the near future.

Fig.1 Internet Bandwidth, 2005- 2015

Figure 1, shows the advancements that have been made in the internet bandwidth over the past five years. It shows that internet bandwidth has increased more than 21 times as now as compared to that of 2005. It is also likely that in response to the current initiatives taken in this regard it will become almost thrice the current bandwidth available to the university, in the year 2010.

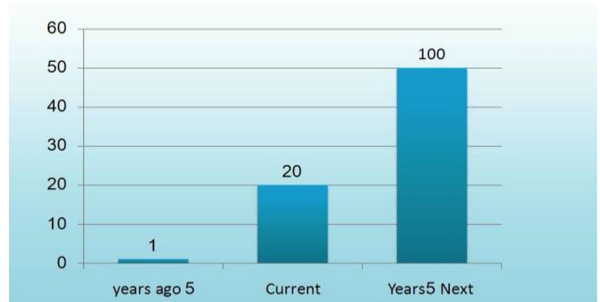

Fig. 2: Electronic Services 2005-2015

Figure 2, shows the advancements that have been made in the availability and usage of e-services over the past five years. It shows that the availability and usage of e-services has increased to 20 times as now as compared to that of 2005. The figure suggests that there was only 1 e-service available in 2005, while now in 2010, 20 e-services are available to be used. It is also likely that in response to the current initiatives taken in this regard it will become almost 5 times more in year 2015 as compared to the current number of e-services available for usage to the university, in the year 2010.

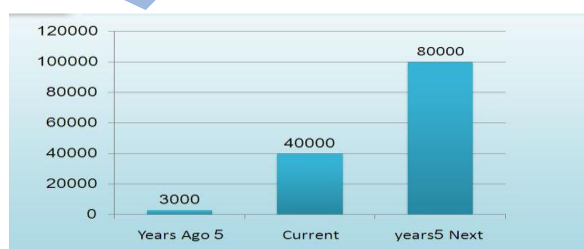

Fig. 3: Number of network nodes 2005-2015

Figure 3, shows the increases that have been made in the availability and usage of network nodes over the past five years. It shows that the availability of network nodes has increased to 13 times as now as compared to that of 2005. The figure suggests that there were only 3000 network nodes in year 2005, while now in 2010, 40000 network nodes are available to be used. It is also likely that in response to the current initiatives taken in this regard it will become almost twice more in year 2015 as compared to the current number of network nodes available for usage to the university, in the year 2010.

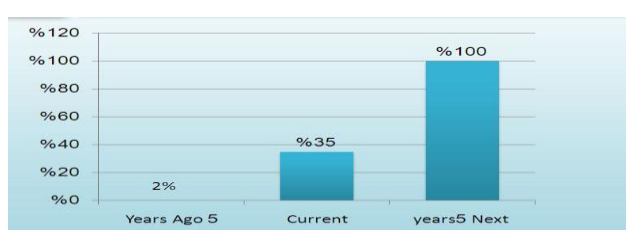

Fig. 4: %age of wireless coverage 2005-2015

Figure 4, shows the advancements that have been made in the wireless coverage over the past five years. It shows that in year 2005 only 2 of the each 100 users had the availability of the wireless coverage, which was just 2% of the total users. The figure suggests in 2010, the wireless coverage has increased to 35%, which is 17 times more than the coverage available in year 2005. It is also likely that in response to the current initiatives taken in this regard it will become almost 3 times more in year 2015 as compared to the current wireless coverage available for usage to the university, in the year 2010.

VI. Emerging Services

While comparing the technological advancements in terms of the network services it has been observed that in 2015, the e-services would be a leading factor and play a very vital role in implementing the KSU's strategic infrastructure. KSU's has not just focused on increasing the number of node, internet bandwidth and wireless coverage, but has also strongly emphasized to make itself as a technology-oriented university, table 1 shows a comparison of the services available in 2005, now and in 2015.

Services	2005	2010	2015
ICT Legacy System	√	x	x
E-Servicelessness	√	x	x
Restricted e-mail access	√	x	x
Client / Server Technology	x	√	√
Technology enabled campus	x	√	√
Distinguished portal	x	√	√
RFID	x	x	√
KSU Channel	x	x	√
Campus wide wireless network	x	x	√
Virtual Campus	x	x	√
LMS and E-Leaning	x	x	√

Table 1: e-services availability comparison

The table 1, elaborates the developments that have taken place in past five years and the expected developments in the year 2015. KSU started with the legacy systems and e-serviceless environment with the e-mail facility only for the faculty. The visionary leadership of the university very soon realized that the e-serviceless environment may increase the data duplication and inconsistent data to be available across the concerned departments. These legacy systems for record management are not being used anymore, and have been replaced by a state of the art enterprise system 'MADAR' which is responsible for the implementation of university-wide financial, managerial and strategic initiatives.KSU, currently enjoys the presence of client server technology to implement most of its procedural needs which is likely to be supported or replaced by the usage of RFID 2015. The distinguished portal of KSU is likely to improve even more in the 2015. Like never before, KSU is set to initiate KSU channel, completely wireless environment, virtual campus and LMS and E-Leaning in year 2015.

VII. Conclusion

It can be concluded that the King Saud University's (KSU) management has very timely identified the establishment and consistency of the e-services across the campus. For that purpose huge emphasis has being given to establish the e-services and enhance the already present services. The university providing suitable funding for that purpose, and as the result of initiatives the internet bandwidth, wireless coverage, e-services usage and network nodes all have grown exponentially to show that the funding is being made in the right direction, and at right time. With the current pace of technological advancement it really looks possible that the university can have its own channel, complete wireless network environment, virtual campus, LMS and E-leaning by year 2015, as decided in the university strategic plan. The deanship of e-transactions and communication is playing a vital role in implementing all such initiatives. It is important to describe that KSU is not only emphasizing in providing the excellent e-services but also has a strong reputation for the highest quality of academic disciplines, academic infrastructure and research attitude. This has been helped the university to be among the first choice of the reputed academician across the globe. King Saud University has been recently ranked at 199^{th} position among the 500 best universities in the world.

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Virtual Backbone Routing Structures in Wireless ad-hoc networks

Longxiang Gao[1] Ming Li[2] Bai Li[3] Wanlei Zhou[4]

***Abstract*-In this paper we studied several virtual routing structures (ring, ring-tree, cord, and mesh) which are scalable, independent on addresses, based on local information and partialglobal information for routing packets. These virtual routingstructures are built on the top of the backbone nodes whichare selected by considering power, connections, and immobilitymetrics.Our experimental results on the ns2 simulator and both TelosBand MicaZ sensor nodes tested platform prove that the virtualbackbone structures are superior to the existing routing schemesand the different virtual structures fit in with the differentphysical scenarios.**

***Keywords*-virtual routing structure, flat naming, distributed hash table, backbone nodes**

I. Introduction

A wireless ad hoc network is a decentralized wireless network, where nodes can move and reconfigure into arbitrarytopologies. The ad hoc network is different from wirednetwork, where each node is willing to forward data to other nodes based on the network connectivity. Wired network, incontrast, in which routers perform the task of routing. It is also in contrast to managed (infrastructure) networks, in which a special node (server) known as an access point manages communication among other nodes.

There are two fundamental aspects of wireless ad-hoc network that make routing in wireless ad-hoc networks extremely challenging [1]: the mobility character determines that the identities of the mobile nodes cannot rely on the locations or addresses and thus the address-based routings require the extra and expensive identity-location mapping system to support, and the second problem is that the mobile nodes only have the limited resources in terms of memory, power and computing capabilities which determine that the mobile nodes cannot accomodate the big routing tables and cannot undertake the complicated data processing such as the exponontial computing. This paper studies those problems and presents the practical solutions to each problem. The first problem requires that the naming system is independent on the addresses, and has been studied by Mathew [2] and Tsuchiya [3]. Mathew [2] considered the flat-naming to get rid of the addresses and organized a virtual ring structureto route packets, but the adjacent nodes in ring might be far away in the real network. Tsuchiya [3] built a landmark hierarchy which is embedded in the naming of the nodes. Our naming system combines the flat naming idea from Mathew with the hierarchical information obtained from the searching process. The searching system is designed to find the backbone association with the destination node. The second problem requires that the routing process is scalable to the size of the network and routing performance is efficient. Mathew used the virtual ring structure and Tsuchiya and Leong [4] used the tree structure for routing. We analyzedthe different virtual structures (ring, ring-tree, cord, and mesh)based on the backbone nodes. The simulation results show that the virtual backbone routing is workable and efficient. This paper presents the virtual backbone routing structure that offers the flat naming and an efficient routing mechanism. The key important characteristics of the portocol are:

1) The flat naming is applied for nodes identification. Thus the exact physical location is not required, which can be expensive and requires extra location-identity mapping system.
2) The backbone nodes are selected by considering power, connection, and immobility metrics, which abstract the real network topology and mobile nodes capabalities.
3) The flooding or broadcast messages limited in the routing process which can reduce traffic and save power.
4) The different virtual structures have been studied, which show the different advantages and drawbacks in scenarios.
5) The smart routing is demonstrated by the analysis and simulations which show that the routing protocol can take shorter path without going through every node along the route.
6) The dead-end problem can be lessened in which the backbone nodes always lead to a path to the destination node.

The rest of the paper describes the design and simulated performance of this virtual backbone routing scheme. Section2 reviews existing routing schemes in wireless network.Section 3 describes the structure and algorithms of virtual routing scheme in details. Section 4 studies the simulation results. Section 5 suggests areas for future research. Section 6summarizes the paper's contributions.

II. Related Work

There has been a large amount of work on routing protocols for wireless network. Based on their principles and performances,we can classify them into three generations. Eachgeneration has its own characters and contributions.

The first generation of routing schemes for wireless network primarily focuses on data collection. It mainly includes reactiveand proactive protocols [5], [6]. The proactive protocols,such as Destination Sequenced Distance Vector

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(DSDV) [6],[7] and Optimized Link State Routing protocol (OLSR) [8],need to maintain consistent up-to-date routing informationfrom each node to every other nodes. Reactive protocols,such as Dynamic Source Routing (DSR) [9], [10] and AdhocOn-demand Distance Vector Routing (AODV) [11], [12],construct routes to destinations only when they need. The
need is initiated by the source, as the name suggests. When a node requires a route to a destination, it floods a route request message within the network. If a node receives a route request, it responds with a route reply. Unfortunately, this process may lead to significant flooding the network in order to discover the desired route. The first generation wireless network engrosses vast of bandwidth and energy due to the flooding traffic involved. Also it is difficult to be implemented in a large scale network. The second generation of routing protocol for wireless network is the coordinator-based routing protocol which uses coordinator-based identifier [13], [14], [15] to locate nodes and perform the searching and routing process. Beacon Vector Routing protocol (BVR) [14] and Greedy Perimeter Stateless Routing (GPSR) [15] belong to this category. These schemes need the global or local coordinators to direct the routing process. The coordinator-based routing protocols are efficient and smart, since they utilize the geographic information to get the accurate address of the destination and use this information to forward packets. However, in order to implement the coordinator-based searching and routing functions, we need to set up the fixed infrastructures for the servers [15] or beacons [14] and maintain them as well. That could be very expensive or impossible in some cases to equip every node with the global server (e.g., Global Positioning System) [15]. Further, these local or global infrastructures need to be updated regularly.
During this time, a new peer-to-peer mapping and searching protocol, Distributed Hash Table (DHT) [16], [17], [18], [19], has been developed. DHT is a building block for peer-topeer applications. At the basic level, it allows a group of
distributed hosts to collectively manage a mapping from keys to data values, without and fixed hierarchy, and with very little human assistance. This building block can then be used to ease the implementation of a diverse variety of peer-to-peer [20], [21] applications such as file sharing services [22], [23], DNS replacements [24], [25] and web caches [26], [27]. Although, DHT originally was created and used in wired network, ideas and mechanisms can be applied and implemented in a wireless network. With the DHT, the third generation of wireless network with name routing becomes true. Each node can use a name as a unique identifier (i.e., MAC address) and distribute it among in a wireless network without pre-defined address information. By using this way, name routing avoids drawbacks of address routing and can be deployed in a large network where each node can identify its name by through self-learning process. Virtual Ring Routing (VRR) [2] is an originator for the name routing. The VRR uses the node's natural identifier and DHT to map nodes into a ring. Based on this ring, VRR performs continuous hash searching and single route routing. This routing protocol can achieve name routing and avoid drawbacks in the address routing. However, the routing performance of VRR is in doubt. It uses clockwise or anti-clockwise to forward Node Hashed Identifier packets, sometimes this single route may be the worst route. And also, the VRR has a high risk to encounter the dead-end problem [28], since the DHT ring is created by continuous hash function and with
the order of nodes' hash value. Recently, we developed the virtual backbone routing scheme which belongs to the name routing category, but it improves the routing performance
which VRR has suffered from the single level routing.With the associated backbone nodes, normal nodes in this structure can maintain their natural connection relationship and find other nodes' relationships, which can efficiently mitigate the deadend problem. With the virtual backbone structure, packets do not need to go through every node along the route to the destination. Instead, lots of packets can be forward direct to their destination with the backbone node's directer. There are some challenges to implement the name routing in wireless networks: addresses can not be used to identify mobile nodes because network topologies are not static; some wireless node has limited resources in terms of memory and power which are sensitive to the size of the routing table; and the routing should be efficient in terms of hops and power. The virtual backbone routing scheme aims at fixing these problems. First, it gets rid of address altogether. That is, we propose to route directly on the name identifiers of the wireless node. Secondly, with backbone nodes, rest of normal nodes can be associated. Thirdly, normal nodes can use the backup route provided by their associated backbone node to route packets and avoid the dead-end problem in some certain extent. Finally, we can organize those backbone nodes into a virtual structure, such as Cord[29] and Ring[2], and use the Distributed Hash Table (DHT) [16], [18] techniques, to perform the exponential searching process. The main contributation of this paper is to introduce and implement the backbone idea into the wireless ad-hoc network and organize these backbone nodes into serveral virtual structures. In this section, we introduce how to select and build up this backbone structure. The backbone nodes carry local characters and represent its local nodes. With the natural character of wirless network, such as mobility, fixed structure is not suitable to handle it. Instead, a flexibal virtual structure is suitable to be deployed in the wireless ad-hoc network. We use the following two sections to describe how to select and deploy backbone nodes and how to set up different virtual structure on top of backbone nodes and for different topology:

- Backbone Structure
- Virtual Backbone Routing Structures

III. Backbone structure

The backbone structure is the foundation of this virtual backbone routing scheme. This structure should include several unique backbone nodes. Each of backbone node should represent its local nodes and cover all of its one hop area. The identifier of each node should be calculated based on its own characters rather global identification system, such as IP address. In order to achieve these two design targets, we develop the following two components:

• Backbone Selection
• Naming System

A. Backbone Selection

Backbone selection is designed to distinguish important nodes from normal nodes. Important nodes are used to represent
its local nodes characters which have wide connectivity, more power, and relative immobility. The connectivity means in an one hop physical transaction range how many neighbors it has. The power means how long this node can be used to proces the data with its neighbors. The imobility is used to identify the moving speed of nodes. With these three selection metrices, the relative good performance node can be selectede as backbone node. With the backbone node, rest of normal nodes, which associated with this backbone node, can find the best path to route packet to other nodes which associated with another backbone nodes. Furthermore, with the back-up path provided by the backbone node, normal nodes can mitigate the dead-end problem in the certain extent. In order to select backbone nodes, we mainly have two different ways. One is manual pre-define the backbone node, the other is to use the backbone selection algorithm. In some topology, the network administer can pre-define the network topology and deploy the powerful node as the backbone node. For example, some sensor network is deployed in certain place and record the temperature by using the ad-hoc routing technique. In this network, it can pre-define the location for backbone node during hte design phase. However, this manual pre-define method meets the bottleneck when the topology is dynamic changing, such as wireless mobile network. To solve this problem, we develop the backbone selection algorithm to automatelly get backbone nodes.

The backbone selection algorithm is a self-running algorithm, it does not need to set up any infrastructure. Instead, each node runs this algorithm and gets its backbone selection metric value by itself. Only those nodes whose backbone selection metric values are bigger than the threshold can be selected as the backbone node. The backbone selection metric should include the following items:

• Connectivity
• Power
• Immobility
• Thresold

The connectivity metric is a measure of the relationship among its one hop physical neighbors. Good connectivity means this node can cover a big area and has more chance

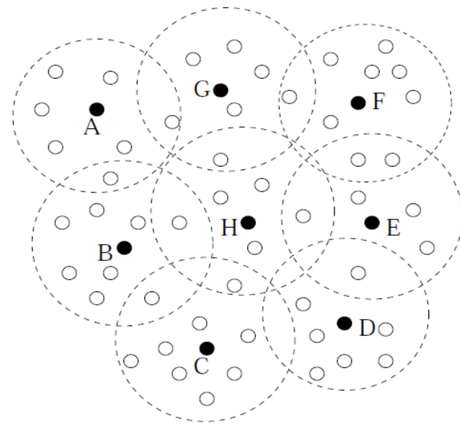

Fig. 1. A Sample of wireless Ad-hoc Network

to be selected as the backbone node. Let C denotes the connectivity metric and N_a denotes the neighbors of node a, including the node itself. Depend on the requirement, we can
set up different base B. If N_a is bigger than the N_B, then the $C=1$. Otherwise, the connectivity metric value C is set up
as the value of N_a/N_B, which should be bigger than 0 and smaller than 1.

The power status is a measure of power capability to represent how long this node is in the activity status. The longer the node is in the activity status, the more likely this node process more data and provide more energy to perform the longer transmation. The power metric P is measured dynamically with a value between 0 and 1, with P_B being a minimally maintained power threshold, if the enery is bigger than P_B, its power metric value P is set up as 1. Otherwise, the
P is set up as P_a/P_B, where the P_a is itself energy. Overall, the power is expressed as a fraction of 1.Tthe higher the value,
the more powerful the node has.

The immobility metric (named as S) reflects the moving speed of a node and is expressed as a fraction of 100. Inthis case, a lower immobility value is more desirable becauseit indicates more stable of the node. So, $1/100$ would be aminimally immobile measure, and $100/100$ is a node that is100 percent unstable.

The selection process is based on Algorithm 1, whichfilters and selects backbone nodes from normal nodes. Atthe beginning, all nodes are treated as normal nodes. Thenevery node runs the Backbone Selection Algorithm 1 and getsthe value of node state. With the node statue, nodes with the$N = 1$ can be selected as the backbone nodes.The selection composite formula consists of values K_1through K_3, known as theselection metric weights. where the$K$ values may be changed according to the real circumstances.By default, K_1 is set to 10, K_2 and K_3 are set to 1. Wed efined the thresholds for the power and the immobility. When

Algorithm 1 Backbone Selection Algorithm

Ensure: $L_{threshold}$: backbone node threshold
Ensure: The connectivity parameter C
Ensure: The power capacity parameter P
Ensure: The immobility parameter S
Ensure: The node states N {0 stands for the normal node and 1 stands for the backbone node}

```
N = 0
calculate selection metrics
Metric(C, P, S) {selection composite formula}
Metric = (K1 * C) * (K2 * P)/(K3 * S)
if Metric >= L_threshold {beyond backbone node threshold} then
    N = 1
end if
```

the power capacity is less than the threshold, the weighted factor, K2 is set to a much smaller value, 0.01. The similar processing for the immobility, when a node moving speed is beyond the threshold, the weighted factor, K3 is set to a largevalue, 100.After the backbone selection process, backbone nodes senda 1 hop association message to inform those normal nodeswho received this message that they can associated with thisbackbone node. Normal nodes send the association acknowledgemessage to the backbone node where the first association message come from and ignore other association Messages. Backbone nodes have a set to store those associated normal nodes who reply the association acknowledge message to the backbone node. As shown in Figure 1, after the automatic backbone selection process, node A, B, C, D ,E, F, G and H are selected as backbone nodes. The signal range of these backbone nodescover all of nodes in this wireless topology.

B. Naming

In our framwork, we have two methods to implement the naming process. One is horizontal naming, the ohter one is the flat naming. The horizontal naming is to use the backbone node identifier to get a new name, which includes both its own identifier and the associated backbone node identifer. After the backbone node selection, every node can find an associated backbone node or if itself is a backbone node, that means it associates with itself. The unique identifier in our
work includes two components. The first component is the associated backbone node, which identify its location area.The second part is its own identifier, which can be used to identify a node in a local area. For example, assume there are six nodes in a local area and with MAC address as their identifiers, namely 4A, EF, B6, 2E, 8A and 3C. After the backbone selection, node EF is selected as the backbone nodes and others are normal nodes associated with node EF. In the naming proces, each node get a unique global identifier automatelly as EF4A, EFEF, EFB6, EF2E, EF8A, EF3C and EF4A.
The flat naming system is a ideal solution for wireless network. Because, this naming system does not need any infrastructure, instead it can use its own natural identifer to identify itself. In this paper, we use the MAC address as nodes' identifiers and implement the flat naming. The MAC address is natural unique in worldwide and has has the same size.

IV. Virtual Backbone Routing Structures

The routing process in our work takes the advantage of virtual structure based on backbone nodes. The virtual structure is more flexible compared with the fixed phycial network. The good flexibly means, nodes organized into it can easy join and leave the network, without change the original network structure. In this section, we introduce four flexible structures can be used to implement the virtual structure based on backbone structures: ring structure, ring-tree structure, cord structure and mesh structure. Backbone nodes are used to represent its local area nodes and reduce the size of virutal structure
and the message overhead, which is essential important for a non-powerful network, such as sensor network. Furthermore, we merge the DHT technique into the design of our routing protocol. Original, the DHT is used as an application in the wired network. Instead, in our case, DHT is used to divide and distribute the global routing information into each backbone node, which clear up the client-server structure in wireless
network.

A. Ring Structure

The ring structure is a flexible and a close-circle, which is suitable to implement the flat naming with DHT technique. In the ring routing scheme, the normal node routes packets to its associated backbone node first and then the backbone node perform the searching and routing functions. If the destination node has the same associated backbone node, packets are directly forward to the destination by through the same backbone node, which takes the advantage of backbone node. Beyond that, the dead-end peoblem can be mitigate insome extent. In backbone nodes, they store a forwarding set,
which is used to record the next hop to forward packets to other backbone nodes. With this forwarding set, backbone nodes can find another path as the back-up to route to the destined backbone node, when the optimum path is dead. If the backbone node itself is dead, a new backbone node is selected by rest of normal nodes automatelly and re-preocess the forwarding set. With these ways, the network reliability can be improved.
The main task for the backbone node is to route packets to the destination. In order to target this task, two routing tables is set up in each backbone node. The local index table is created to record its association normal nodes' informaiton. The global look-up table is created to store part of virtual global routing information and its physical forwarding set. With these two tables, a smart and efficient network routing structure is stand

by.The data in the local index table is recorded during association messages acknowledgement process. When normal

TABLE I
LOCAK INDEX TABLE

Backbone Node Identifier	*Normal Node Identifier*
Node(A)	1, 100, 5
Node(B)	6, 68, 9, 19
Node(C)	8, 78, 21, 29, 41
Node(D)	3, 18
Node(E)	101, 102, 39, 7
Node(F)	206, 22, 23, 13, 16
Node(G)	10, 73, 33, 36
Node(H)	4, 37, 99, 76, 66

TABLE II
THE GLOBAL LOOK UP TABLE FOR BACKBONE NODE A

Backbone Node Identifier	*Virtual Interval of Backbone Node Identifer*	*Physical Forwarding Set*
Node(B)	[B, C)	Set(B)
Node(C)	[C, E)	Set(C)
Node(E)	[E, B)	Set(E)

nodes send the acknowledgement to its associated backbone node, this backbone node record their identifiers into the local index table. For example, as Figure /reffig1 shown, after the backbone selection process, 8 normal nodes are selected as backbone nodes. We use characters A to H to represent backbone nodes and use random number from 1 to 240 to represent rest of normal nodes. The local index table for each backbone node is created as Table I. Each backbone node has different number of associated normal nodes, as shown in this table. Totally, it has 32 normal nodes. The global look-up table is built up with three columns, the first column is the backbone node identifier. The second column is the virtual interval of a range of backbone nodes's identifiers. The third column is the physical forwarding set to its relevent backbone node (the backbone node in the first column). There are logN (N stands for the number of backbone nodes) entries in this table and ordered according the DHT theory [16], [18]. For the backbone node A, it store the entries to the backbone node B, C, E as the Table II. After these process, the virtual routing scheme is built up. Packets can be routed through backbone nodes whenever the

physical forwarding route is set up. The physical forwarding set is used to set up the relationship between two backbone nodes. In order to set up this table, it has two methods. The first method is to go through normal nodes. Normal nodes in our work not only has its own associated backbone node, but also has other normal neighbors which associated with other backbone nodes. Thesenormal nodes get their neighbor's inforamation by regular message exchange and forward this information to their own backbone nodes. Backnode nodes store this normal node as a physical next hop to enter into the another backbone node area. Another method is to set up a different transmation rate. For wireless network a higher transmission rate has a shorter radio range [30], that means if we use a lower transmission rate, we can get a higher radio range. We can set define a

Algorithm 2 Routing Algorithm
Ensure: MyT ype: backbone node or normal node
Ensure: NeighborSet: one hop physical neighbors for normal nodes
Ensure: LocalSet: nodes associated with the backbone node
Ensure: V irtualSet: backbone nodes which in the virtual structure
Ensure: ForwardSet: next physical hop to the different backbone node
1: if The MyT ype is normal node and the destination node is in my NeighborSet then
2: Source node directs route packet to the destination
3: end if
4: if The MyT ype is normal node and the destination node is not in my NeighborSet then
5: Forward this packet to the associated backbone node
6: end if
7: if The MyT ype is backbone node and the destination node is in the LocalSet then
8: Forward the packet to the destination directly
9: end if
10: if The MyT ype is normal node and the last hop is from the associated backbone node then
11: Forward this packet to another normal node which associated with the destination backbone node
12: end if
13: if The MyT ype is backbone node and the destination node is not in the LocalSet then
14: Using the DHT like searching process to look up the V irtualSet locating the next virtual backbone and using the next physical hop from the ForwardSet to forward to its associated normal node
15: end if

lower transmission rate for backbone nodes to transfer data among them in order to bypass some normal nodes. The DHT
like seaching process is using the partial globel information stroed in backbone node to perform the O(logN) look-up performance.

B. Ring-tree Structure

A single virtual ring is flexible but inefficient in term of performance. In a single ring structure, packets are routed unidirectionally, and hence the senders need to check every backbone node along the route. To remedy this shortcoming by diversifying the routes, we introduce a ring-tree structure. The backbone selection in this structure is different from the above one, where it uses the connectivity as selection metric to form the ring-tree structure. Our backbone selection algorithm is autonomous. When wireless nodes are placed into a network, they run the backbone selection algorithm to exchange the information of the physical connections. After several rounds' selection, every node will eventually receive its virtual level information. We show this convergence in thefollowing example. In a graph G = (V,E), a vertex v 2 V

represents a node, and an edge (u, v) indicate a link connection that two nodes u and v are in each other's transmission range. A subset of the vertex, V_b _ V , is a backbone set if each vertex in $V - V_b$ has at least one neighbor in V_b. A virtual backbone ring is constructed by virtually connecting the backbone nodes
(See Algorithm 3).

Algorithm 3 Ring-tree Backbone Selection Algorithm

```
int PreLevel=1, MyLevel=0;
Receive backbone selection message with the neighbor set parameter
while MyLevel==0 do
if my neighbour set includes the neighbour set then
Inform source sensor node to remove itself, and increase PreLevel by 1
else
MyLevel = PreLevel
end if
end while
```

In the first step of our backbone selection algorithm, every normal node sends hello message (HelloMessage()) to all of its 1-hop physical neighbors, where the HelloMessage() includes the source node's MAC address. When a receiver receives the HelloMessage(), it retrieves the sender's MAC address from the packet header and stores this MAC address to its own physical neighbor set. After a while, every node eventually obtains all its physical neighbors' MAC addresses and stores these addresses into its own physical neighbor set. The second step of backbone selection algorithm is to send the backbone selection message. The backbone selection message includes the current physical neighbor set for each node. When a node receives backbone selection messages, it refers to the source node's physical neighbor set and compare it with its own physical neighbor set. If the receiver's physical neighbor set includes the sender's physical neighbor set, then we define the receiver as a key node. Then, the receiver sends an eliminating message to the sender and the sender assigns the selection round number as its virtual level information—if the round number is 1 then the virtual level information for the sender is also 1. After this initial phase, the receiver will enter into the next round of the backbone selection. If the sender's physical neighbor set includes the receiver's physical neighbor set, then the receiver removes itself from the set and informs the sender to commit int the next round. (The receiver assigns a round number as its virtual level information during this phase.) If neither the sender nor the receiver is included in each other's neighbor set, then this backbone selection message is dropped so that both nodes enter the second round. In the second round, each node updates its "neighbor" set, which includes the alive neighbors survived after the first round elimination. After several rounds, all nodes will be eliminated.
Eventually, only one or zero node will survive in this sensor network — the backbone selection process terminates when no node is left; and the process also terminates when a single node remains, so we simply assign the current round number

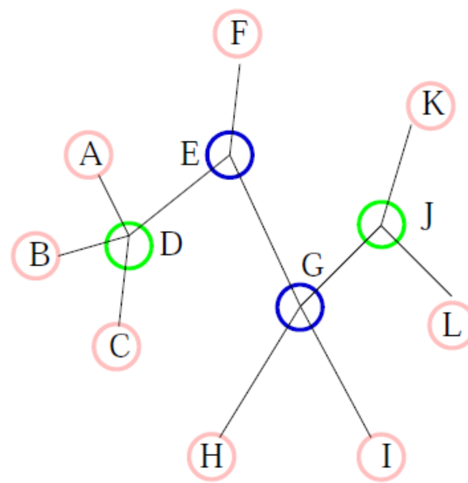

Fig. 2. A sample of wireless network

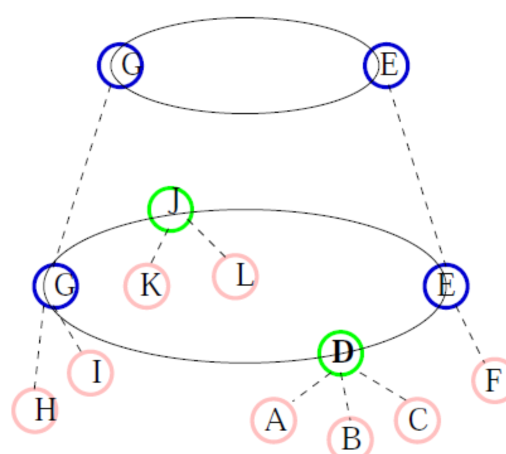

Fig. 3. After the backbone selection process, the position of each node on the ring-tree structure

to this node's virtual level information. Until now, every node obtains a piece of the virtual level information which is used in the following processes.
Figure 2 gives an example for executing the selection process in a wireless network. In the first round, the red nodes are grouped into the "leaves" nodes as they are the "children" of the green nodes and the blue nodes. Then, the red nodes are eliminated from the first round of backbone selection. So, the green nodes and the blue nodes form a new level 1 backbone ring. In the second round, the green nodes will be removed because they are the "children" of

the blue nodes. So, the blue nodes form a level 2 backbone ring. In our case, nodes A, B and C are children of D; nodes H and I are children of node G; nodes K and L are children of J, and node F is the child of E. The key nodes G, J, D and E form the bottom virtual ring; nodes G and E form the second virtual ring which is mapped to the bottom virtual ring, as shown in Figure 3. The routing process in ring-tree structure inherits the advantages of multi-level virtual ring [31], where packets are forwarded efficiently to the destination without flooding. After the searching process, the source node can obtain the destination

level. The wireless node compares the cross-level interface with its own physical neighbor set, and therefor determines which physical neighbor to be the most suitable node for the next hop.Every node in ring-tree structure maintains a routing table where stores the virtual level interface and the physical neighbor set. When a node receives a packet, it compares the destination level information from the packet header with its own level by executing our level similarity decision algorithm as in Algorithm 4. If the destination level is higher than its own level, then it will look for a up-layer interface. For example, in the Figure 3, the up-layer interfaces for nodes D,E,G and J are G and E. The next step is to compare these interfaces with its physical neighbor set. If the algorithm finds one fromits own physical neighbor set which is equal to one of the interfaces' set, then it will choose this node as the next hop; Otherwise, it will forward this packet to a random physical neighbor which will use the level similarity decision algorithm to continue the process.

Algorithm 4 Level Similarity Decision Algorithm
1: Receive forward packet with dest level and dest identifier parameters
2: **if** The dest level > my level **then**
3: Source node looks up its upper interface and compares with its physical neighbor set to find which one should be selected as next hop
4: **else if** The dest level < my level **then**
5: Source node looks up its down interface and compares with its physical neighbor set to find which one should be selected as next hop
6: **else**
7: Packet has been arrived on the destination level and should forward this packet to its next hop according its virtual neighbor set
8: **end if**

Each node in ring-tree structure uses this level similarity decision algorithm and calculates its next hop towards the destination level. Eventually, the packet will be forwarded tothe destination level. When a node receives a packet and finds

the destination level is the same as its own level, it will forward the packet to one of its virtual neighbors in a clockwise or anti-clockwise ring similar to VRR [2]. Our idea behind MVR routing forward is to take advantages of the cross-level forwarding, so that packets can bypass some "useless" routing nodes and take the shortcut path in a crossing level.

C. Other Structures

Cord and mesh are another two virtual structures which can be used to organize backbone nodes and perform routing function. Virtual cord routing structure is to organize these backbone nodes into a line and treat their associated noraml nodes as leaves. The routing for the virtual cord structure is to perform the one way searching and routing, where every backbone nodes are mapped to a single virtual line.

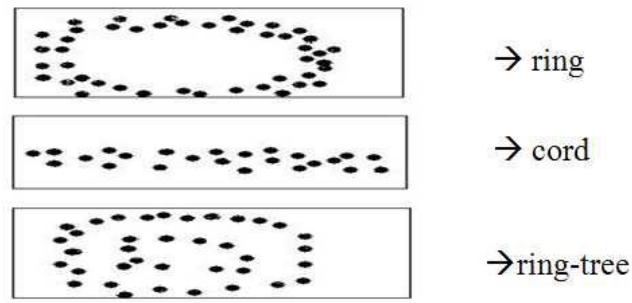

Fig. 4. Different Virtual Structure for Different Physical Topology

Compared with ring and ring-tree structures, cord is easy to be set up and maintain. However its routing performance is a big concern. The mesh structure is hard to build, but the routing performance of this virtual structure is efficient. The routing in the mesh structure is point to point. That means, backbone nodes can directly route packets among themselves. We believe different physical topology should be organized into different virtual structure, as shown in Figure 4.

D. Scalability

One of the concerns for a routing mechanism is how to make it scalable for a large network. Future wireless network[32] should be a network where a node can join and leave easily without affecting the structure and connections with other nodes. The virtual backbone routing performs well in terms of scalability. For normal nodes, they can easily join into a network by find an associated backbone node. When they want to leave the network, they only need to broadcast an one hop message to inform its neighbors, including the associated backbone node, that this node is going to leave the network. For backbone nodes, from backbone node selection critical, we use relative immobile and powerful nodes as backbone nodes. If it does need to leave, we can use the backbone selection algorithm to re-select a new backbone node to take over the original backbone node function automatically. In some network, such as the wireless sensor network, the memory size of each node is limited. So, the size of content in a node becomes a big concern for a large scale network.In our work, the size of content in a normal node is much smaller compared with the backbone node. And even for the size of content in backbone node, the difference between a small network (less than 100 nodes) and a large network (more than $1,000$ nodes) is roughly constant. Clearly, we set

10 bits node identifiers, which can express more than 1, 000identifiers. For the local index table, assume each backbone node store 50 normal nodes' identifiers, that means it stores about 625 bytes. In the global look-up table, for each entry it needs 10 bits to store the backbone node identifier, 2*10 bits to store the virtual interval of backbone node identifiers and roughly 10*10 bits for the physical forwarding set, assuming each forwarding set has 10 items.

So for each entry, it needs about 16.25 bytes in memory and there has $\log_2(1000)$ entries, that means it needs 162.5 bytes for the global look-up table in total. These two table take less than 1K space in memory and is much smaller than the sensor node memory, which is equipped with more than 4K memory [33].

V. Evaluation

We evaluated our work using both simulations on *ns-2*[34] and physical measurement on a tested sensor network — a 30 TelOSB node sensor network[35] and 10 MicaZ node sensor network. Since the virtual backbone routing structures(VBS) are new virtual routing structure and the routing performance of ring-tree is better than the VBS (ring-tree uses cross-level routing forward mechanism), we compare VBS with virtualring routing (VRR) [2]. However, VRR has copyright from Microsoft and we are not entitled to port the source code to *ns-2*. We create a single virtual ring routing protocol (SVR) to represent the routing function provided by VRR. Furthermore, we compare AODV and DSR, which are representative wireless routing protocols with well tuned implementations in *ns-2*. We implement and test our routing protocol on the MicaZ platform. Our testing results show that VS performs well across all the experiments comparing with SVR which performs worse than tractional routing protocols.

A. Simulations

In this section, we simulate our work with other well-know routing protocols. The simulation experiments run on *ns-2* simulator [36], using the wireless extensions developed by the CMU [37]. This simulation environment offers high fidelity, as it includes full simulation of the five layer TCP model[38].

1) Experimental Setup: Since in this simulation we mainly focus on the newtork layer, we set up the same parameters on the other four TCP model layer. In the applicaiton layer, weuse the Constant Bit Rate (CBR) as the data traffic and UDP as transport layer protocols to test the seaching and packet delivery performance. For the data link layer and physical layer protocols, we just use the normal 802.11 environment. In this simulation, We random setup 400 wireless nodes in a 1000 X 1000 plane with the signal transmission as 40 meters. Furthermore, each node has a random speed in the interval [0, 10] m/s.

2) Evaluation Metrics: In order to evaluate its performance, the following aspects have been studied:

1) Searching performance: the number of hops it takes to find the required node.
2) End-to-end delay: the total time it takes from a source node to the destination node.
3) Routing efficiency: in this experiment, we measure the number of hops it in different routing schemes.
4) Message overhead: this is the total number of control messages generated to exchange and maintain routing schemes.

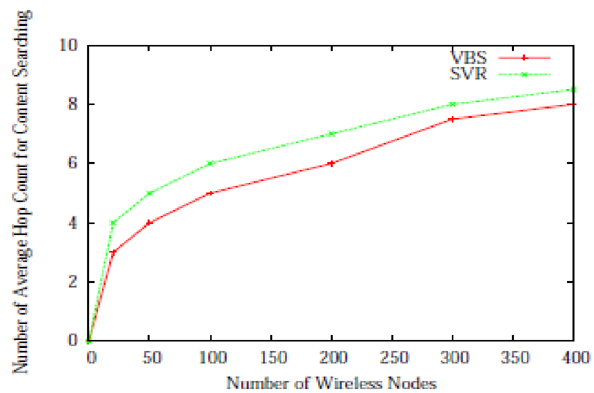

Fig. 5. Hop Counts to Find a Certain Content

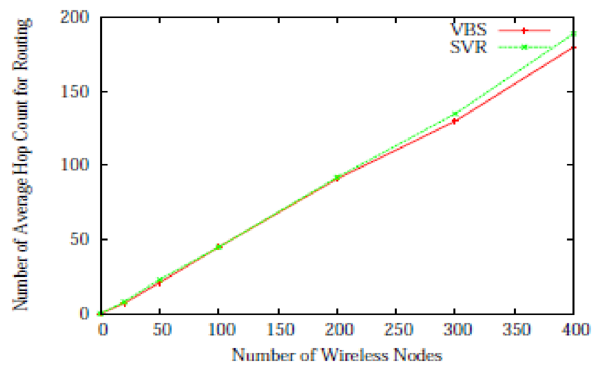

Fig. 6. Routing performance

3) Performance of searching a target node: One of purposes of this new routing scheme is to quickly find and locate the target node. We use the embedded DHT technique to organize all of backbone nodes into a virtual structure and perform the $O(\log N)$ searching process. In this experiment,

we test its performance of searching the target node by calculating how many hops are needed in a dynamic network.We compare its searching performance with the SVR. The searching performance has been tested in the different size of wireless networks which have different number of nodes, range from 50 to 400 respectively. Results are shown in the Figure 5.

Based on these results, we see its searching performance is roughly constant to the size of the network, which indirectly prove the embedded DHT technique is suitable for our architecture. By analyzing these results, we can find the searching process needs only about O(logN) hops. where the N stands for the number of backbone nodes in a wireless network. While in the Single-level Virtual Ring (SVR) structure, the searching process uses the one-way searching, either clockwise or anti-clockwise, thus the performance of SVR is sensitive to the size of the network in which the number of hops required for searching is proportional to the size of the network.

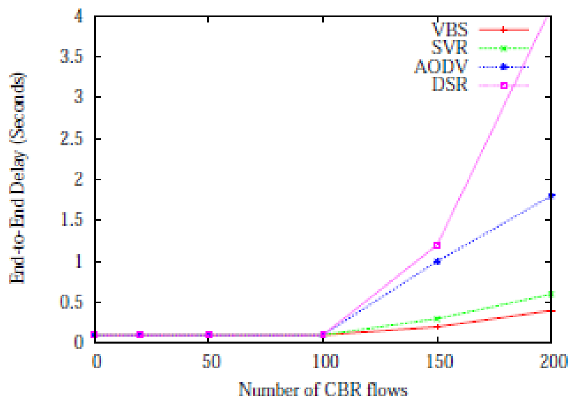

Fig. 7. End-to-end Delay

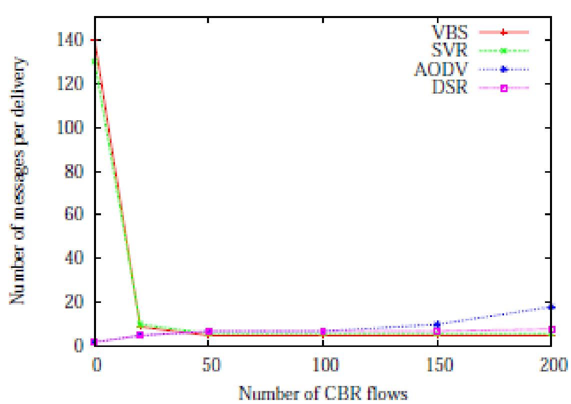

Fig. 8. Message Overhead

4) *Performance of routing efficiency*: Another purpose of this routing scheme is to efficient route packets to the right destination. We designed experiments to test how many hops are needed to route packets by using the virtual backbone routing algorithm and comparing with using the SVR routing algorithm. We have tested wireless networks in different sizes, range from 50 wireless nodes to 400 sensor nodes. The results are shown in the Figure 6. Based on these results, we can see our routing performance is slightly better than the SVR. This is because both of these routing schemes include virtual structure and using part of DHT techniques. However, in a small area, our architecture is better than the SVR. Because, we use the backbone node as the co-coordinator in a small area. When the source node and the destination node share the same backbone node, it does not need to go through other backbone nodes, instead it can route to the destination directly. While for SVR, it still need to perform the full routing selection process, whcih cannot take this bypass route. And also, it shows that, this technique is much useful for date management in a local area, such as battlefield.

5) *Performance of end-to-end delay:* The end-to-end delay is an important concern in wireless network. If end-to-end delay is high, this routing protocol is unacceptable. We design an experiment to test the performance of end-to-end delay. The experiment tested the CBR traffic, range from 50 CBR flows to 200 CBR flows. The source node and the destination node are randomly chosen and we measure their average performance. From the Figure 7, we can see the performance of end-toend delay for our work is similar to AODV and DSR, but it is slightly better than the SVR. The reason is that the SVR just maps all nodes' name identifiers to the single ring and applies the one-way route to forward, where it is mapped by the order of nodes' hash value and uses the clockwise to route. Since, this routing protocol is likely to encounter the dead end problem, SVR needs time to process these dead end fallback. Instead, lots of CBR traffic can be routed in the local area with the backbone node in our work, where packets do not need DHT technique to perform outsider searching process. By using this way, packets can bypass some "useless" node so that the end-to-end delay could be reduced accordingly. This idea is similar to the address routing but we implement it on the content routing structure.

6) *Message overhead:* To study the efficiency of a routing protocol, we analyzed the message overhead required for our date management architecture. The number of CBR traffic message is the same as before, ranges from 50 to 400. From the Figure 8, we can see the messages overheads for the AODV and the DSR are quite close, which both start from 0. The result show our work and the SVR need more overhead messages at the beginning. After the virtual structure being
]built up, our architecture needs the least overhead packets to finish a delivery. However, for the SVR, it needs much moreexchanges messages to finish a packet delivery compared with other three routing protocols. This is because the backbone based virtual routing scheme needs to run the backbone selection algorithm at the beginning to select backbone nodes. After the backbone selection process, our work needs the least message overhead to route packets, while the SVR uses less message exchanges at the beginning, but it costs more to maintain and repair the dead end problem.

B. Sensor Network Testbed

We work on the open sourced TelosB [39] research platform. We assume a small (several cubic inch) sensor/actuator unit with a CPU, power source, radio, and several sensing elements. The processor is a 8 MHz TI MSP430 microcontroller with 16 KB of instruction memory, 10 KB of RAM for data, and 48 KB of flash memory. The CPU consumes 1.8 mA (at 3 volts) when active, and 5.1uA power when sleeping. The radio is a 2400 MHz to 2483 MHz globally compatible ISM band, delivering up to 250 Kbpshigh data radio bandwidth on a single shared channel and with a range of up to a few dozen meters or so. The RFM radio consumes 4.8 mA (at 3 volts) in receive mode,

TABLE III
PERFORMANCE OF THE ROUTING ALGORITHM WITH DIFFERENT NETWORK SIZE

Network Size	*Assocating Time (secs)*	*Std. Dev.*	*Routing Time (secs)*	*Std. Dev.*
10 nodes	1.78	0.09	12.75	0.14
20 nodes	1.93	0.12	13.28	0.16
30 nodes	1.45	0.14	13.94	0.15

TABLE IV
MEMORY USAGE ON OUR ROUTING PROTOCOL

Mote Types	*Standard RAM (bytes)*	*Used RAM (bytes)*	*Standard ROM (bytes)*	*Used ROM (bytes)*
TelosB	10K	3238	48K bytes	38682
MicaZ	4K	3354	128K bytes	34136

set to 1 m and the radio transmission range of each node to approximately 1.5 m. The proposed routing mechanism was implemented on a TinyOS v2.1 / TelosB programming platform with 802.15.4 MAC feature in order to incorporate the IEEE standards as well as the efficient traffic management. 4 backbone nodes were pre-defined at each run. We also randomized backbone node selection and node placement to maintain the experimental validity and efficiency. In our experiments, we ran 15 tests for each network size. Both source node and destination node (backbone node or normal node) were placed on the edge of the network in order to get the maximum routing execution time with two clusters inbetween. This setup makes sure the routing message will go through all the clusters in order to reach the destination node.

2) Experiments: The focus of the experiments was to testthe performance of the routing and program compatibilitywith respect to effective and efficient routing, while at the same time maintaining best selective paths. We assessed the performance of our method by measuring the following performance metrics:

• Average associating time: Average time for associating to be completed; the associating time for each run is calculated from the time the nodes started functioning to all the nodes are included in a cluster.

• Average routing time: Average time for routing to be completed; the associating time for each run is calculated from the time the source node transmitted a message to the time that the destination node receives the message.

• RAM size used

• ROM size used

From the Table III, we can find that the performance of the routing algorithm. In terms of number of different network size, the average execution time can slightly increase with the node number rises. It is reasonable when the network

up to 12 mA in transmit mode, and 5A in sleep mode. The whole device is powered by two AA batteries.

1) Experiments Setup: We have implemented a WSN testbed with Crossbow's TelosB motes. In our testbed, up to 20 sensor nodes were located on a regular grid of 10 (5x2), 15 (5x3), and 20 (5x4). The closest distance between nodes was

gets more complicated. Baed on the Table IV, we can find find the performance of the memory usage on our protocol running on different platforms. TelosB spends less RAM and more ROM than Micaz [35] as indicated. Since the byte consumption is still not beyond the manufactured hardware capacities, there will be no compatibility for TelosB and MicaZ to run our protocol. This is particularly useful to incorporate our protocol in applications requiring small memory usage.

VI. FUTURE WORK

Our work is a new name routing in wireless ad-hoc network. It overcomes the drawbacks of address routing and improves the routing performance of the VRR name routing. With the natural advantages of the backbone co-coordinator, the following area can be extended:

• Virtual backbone architecture can be used in a large wireless network with several backbone domain. In future, every backbone with its associated normal nodes can be treated as a domain, which is similar to the local area network (LAN) structure in wired network. The backbone node is the interface in its own domain and organizes other normal nodes work together to perform the network address translation (NAT) function. This is because, the backbone node stores other normal nodes' information. So, they can be used to implement NAT function to translate nodes' name from inside to outside and from outside to inside as well.

• This virtual backbone architecture can also carry out the grid computing and cloud computing. With the natural characters of backbone architecture, data can be easily distributed [40] among the backbone domain to achieve the load balancing. When it needs to process complex computing, nodes who associated with the backbone node can be organized as a single processing point. Complex data or function can be divided into multiple small data units and each unit can be distributed to some nodes, which depends on that node's capability. Backbone architecture is natural suitable for data distribution and load balancing. Further, Virtual backbone architecture can be developed as a energy efficient routing protocol.

VII. CONCLUSION

We present a new virtual routing structure based on backbone nodes, which gets rid of addresses in routing. With the backbone selection and naming, the virtual backbone routing structure can be built up. With exponential searching algorithm and efficient routing forward, our virtual structure achieves a good routing performance. The backbone selection algorithm was designed to abstract the physical connections among nodes in which some "important" nodes are selected as backbone nodes. The

embedded DHT techniques are used to distribute the node identifiers among backbone nodes tom improves the searching performance. With the simulation results, we prove these virtual backbone
routing structures can be implemented and it is suitable for the next generation wireless network compared with other address routing protocols. However, it still can not root up the dead end problem and has some concerns (i.e., high overhead at the beginning) worth further investigation in the future research work.

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Pc Access Control Using Voice Authentication

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Abstract- **Biometrics is a secure way of carrying out access control authentication as it makes use of a person's characteristics. Voice authentication, a form of biometrics, has become more popular and more accepted in the development of commercial applications. This makes it acceptable as an alternative or complement to other forms of biometric authentication. A lot of voice authentication software is developed for companies that use it for telephony operations; this makes it difficult for users who want to use voice authentication for personal purposes to afford such software because companies can afford to pay much more than individuals. This paper gives an overview of biometrics, voice authentication, and then discusses the development of a system that uses voice authentication to access computers as an alternative to other forms of biometric access for computers such as fingerprint readers.**

I. Introduction

The use of voice authentication for access control systems is classified under biometrics. Biometrics is using automated methods to identify a person or verify the identity of a person based on a physiological or behavioural characteristic. Common physical characteristics include fingerprints, retina characteristics, hand or palm geometry, facial features and iris characteristics. Behavioural characteristics, which are traits that are learned or acquired, include signature verification, voice verification and keystroke dynamics. The problem that biometric person authentication deals with, can be summed up as follows: given some physiological or behavioural characteristics of a subject, the so-called biometrics, and those of a reference person, whose identity is claimed by the subject, confirm or deny the claimed identity (Dugelay et al, 2002). Biometric authentication is preferable to any other kind of authentication, such as password, PIN, smartcard, etc, because it cannot be borrowed stolen or forgotten. In addition, it is virtually impossible to forge a biometric (Liu & Silverman, 2001).

Out of the behavioural characteristics mentioned, technologies for signature and voice are the most developed. When considering factors such as accuracy, ease of use and user acceptance, voice gives the overall best benefit (Greene, 2001) An advantage of voice authentication is that it permits remote authentication, unlike other biometric approaches

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such as fingerprint or iris scans, that is, a user can enrol in and work with a voice-authentication system from a remote location such as a telephone. Also many users of biometric systems see fingerprint or iris scans as invasive, they are more comfortable identifying themselves by speaking (Vaughan-Nichols, 2004).

Another advantage is that voice biometrics is cost effective to implement since it can easily be integrated with an existing authentication infrastructure and require no sophisticated equipment (Li et al, 2000).

II. How Voice Authentication Works

Voice authentication systems capture and digitize speakers' voices. The basic equipment is a microphone or telephone to input speech, an analog-to-digital converter to digitize the spoken words, a high-powered computer, and a database to store voice characteristics (Vaughan-Nichols, 2004).

According to Dugelay et al (2002), there are three phases in the process of voice authentication. These are

1. Enrolment: The rightful system user registers a voiceprint to the system.
2. Test: The claimant speaks to the system. The system either accepts that the claimant is the rightful system user, or rejects the claim.
3. Adaptation (optional): When the system decides that the rightful system user has spoken to it, it updates its model of the rightful system user.

During enrolment, the amount of data used determines the performance of the application using voice authentication. For example, the system user(s) might speak a particular phrase or sentence which will contain as many low pitch phonemes – a language's smallest distinctive sounds – as possible, because these are less susceptible to change; or the user might be required to speak the registering phrase a number of times, allowing the system to construct a template made up of a range of voiceprints (Currie, 2003). Another enrolment approach is to create a voice template using a number of different registration phrases. This allows random phrases to be used during authentication.

The voiceprint created during enrolment is then stored as a digital file in a database. The system does not give a yes or no answer during the test phase, but calculates a probability score that indicates how closely the spoken voice matches the stored voiceprint for the person the speaker claims to be (Vaughan-Nichols, 2004). The score is calculated from measures which take into account the statistical distributions for a particular speaker, the content of the message, and information about the environment and recording medium (Bonastre et al, 2003). The general measures that can be used to calculate a voiceprint score are Template Matching and Feature Analysis. Template Matching attempts to work out the probability that one voice print is the same as another voice print based on comparisons of the amplitude of the voice signal at various frequencies at various times really use any characteristic of speech, but has more to do the way sounds change into one another both inside and between phonemes. It digitises human speech and subjects it to certain mathematical techniques so as to reduce it to a series of mathematical values which can then be analysed (Currie, 2003).

The score obtained is then compared to a decided threshold. The threshold is determined while bearing in mind the false acceptance rate (FAR) and false rejection rate (FRR). The FAR is the frequency with which the system accepts an impostor, and the FRR is the frequency with which the system rejects a valid user. If the threshold is too low, FRR will reduces, but FAR will be high. If the threshold is too high, vice versa is the case; the error rate at the point when FAR equals FRR usually gives a good idea of the accuracy and reliability of the authentication system. Depending on how the score compares with the threshold, the system decides on whether to accept or reject the user.

III. Motivation For Study

There are many ways in which biometrics has been used to make authentication easier, especially in access control. A well-known application is the use of fingerprint readers for logging into computers. This study looks at providing an alternative to those who are still uncomfortable with such technology, or who are opposed to it. It is especially motivated by the fact that most voice technologies out there are aimed at corporations and not the average computer user.

Using voice authentication to access a PC makes use of a less intrusive form of biometrics, yet is still more secure than using a password. Since most computers have sound cards that include a microphone and speakers, only the software for voice authentications access needs to be installed for full functionality. In addition many voice authentication technologies available are geared towards telephony making it easily possible for a PC using voice authentication to be used for remote login.

IV. Related Works

Carnegie Mellon University has a research project which came up with the speaker recognition toolkit, Sphinx that contains libraries which can be used in the development of a system implementing voice recognition. It has acoustic model training, comes with a public-domain pronunciation dictionary, and is capable of speaker adaptation.

Summerfield et al write about Centrelink, a company in Australia, which uses voice authentication to verify its customer database. It allows its customers to carry out transactions via telephone using the customer database to over the entire period of the authentication phrase. Feature Analysis (sometimes called Feature Extraction) does not verify identity. Their system also allows customers to enrol for the voice authentication service over the phone. The Australian government also uses them to authenticate welfare recipients.

Petry et al (2008) propose a system for logging into a computer network using speaker authentication. They use a client/server model, where the voiceprints are stored on the server-side. Client and server communicate via the internet and the traffic is encrypted to ensure privacy of the communication.

Voice Verified, a company specializing in voice technology, offers solutions for companies which want to implement voice authentication and/or recognition in their business plan; however, it is not geared towards the average computer user.

V. Proposed System

The proposed system would make use of voice authentication as either the primary or secondary means of access into a computer. It would be developed with the average computer user in mind who would prefer another alternative to the current biometric technologies available, such as fingerprint readers. Since it is meant for use on a single computer, it would not require communication with any external source. As this system is developed with the average personal computer user in mind, the issue of storage is not really a factor. A limit of five users can safely be set to limit the amount of storage required on the computer. To satisfy privacy concerns, only the voice features will be stored.

The enrolment process would involve each user slowly and distinctly listing the numbers zero through nine as prompted by the software. The features from the voice are then extracted to form the voiceprint and stored.

For authentication, the system will generate a random sequence of numbers. The length of the sequence would have been determined by the user during the enrolment phase. Once the user says the numbers, the system will not only try to match the voice to a registered user in the database, but also ensures that the user is saying the numbers given. This eases fears of an imposter recording a legitimate user's voice and using such a recording to gain access to the system. A probability score is calculated after comparing the user's response to the voiceprint stored in the database. If the score is above a predetermined threshold then access is granted to the user, otherwise, the user is denied access. No matter what decision is taken, the system will keep track of all attempts to gain access to the system. The procedure described is illustrated in figure 1.

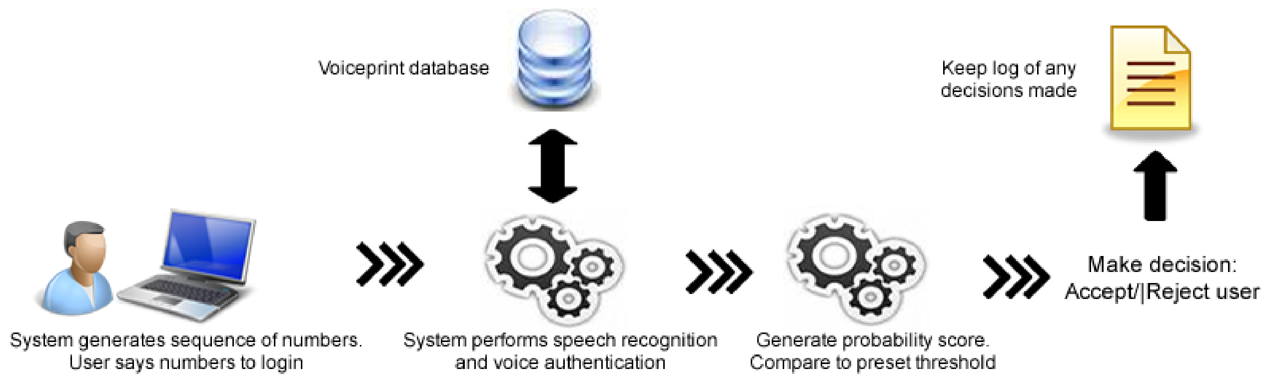

Fig 1: Illustration of proposed system

VI. Challenges Still To Be Overcome

Voice authentication systems usually require chips that can quickly process the large amounts of information involved, in addition to systems with huge memories to store the data and pattern-matching technologies to compare live speech with stored voiceprints. Without this, the time required to verify a customer can be quite long because voice templates are so much larger than other kinds of biometric information. For example, data associated with a fingerprint may take up only 10 Kbytes, while a voiceprint typically takes up from 500 Kbytes to 1 Mbyte. This makes fast database servers and quick filtering software a must.

While it is possible for someone could play a recording of someone's voice to fool a voice authentication system, more sophisticated systems create detailed voiceprints that should not readily match with a recorded voice. Skilled human imitators, though, could still fool a pure voice-authentication system in many cases (Vaughan-Nichols, 2004).

The accuracy of voice authentication systems can be affected by background noises (Liu and Silverman, 2001), or changes to the user's voice caused by illness or stress.

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Understanding Mental Sicknesses Through A Concept Map

Goutam Kumar Saha

***Abstract- This* article aims to visually describe the important concepts of mental sicknesses and mental hygiene and the relationships thereof using a Concept Map. Such knowledge representation technique eases the tasks of managing large representations for complex domains and sharing knowledge with peers and colleagues and publishing them. Mental Hygiene is a very important domain to prevent our mental sicknesses in this modern ultra busy life. The concept map here would be of immense help for faster, easier and effective understanding of the complex Mental Hygiene domain without reading large texts lines after lines and pages after pages. Professional hazards and stringent product or service delivery schedules make our lives full of stresses resulting in mental defects followed by irreparable damages in our bodies. Readers would clearly understand the ways to avoid mental illnesses and to prevent allied diseases on going through the concept map on mental sicknesses and mental hygiene. IHMC's information technology enabled CMap tool has been used here to develop the concept map. Everybody including scientists, engineers, information technologists and other professionals must be aware of this knowledge domain in order to keep their feelings, thoughts and reasoning defect less and to have robust mind. This work is a significant step forward toward development of Concept-Map based Public Health Informatics.**

Keywords: Cognitive Engineering, Concept Map, E-Learning, Mental Health

I. Introduction

Most of us often neglect our mental health issues. This article aims to help us quick understanding of the complex but important mental health issues. Mental hygiene is the science of keeping the mind, brain and nerves of humans healthy. This is concerned with the prevention of mental sicknesses and maintenance of health and psychological issues like perception, cognition etc. Medicine helps biology whereas mental hygiene movement might help us to take care of human psychology. Mental Hygiene is a very important domain to prevent our mental sicknesses in this modern busy life to overcome stress, profound boredom, mental fatigues etc. Cognition is the psychological result of perception, learning and reasoning. Perception refers to the state of becoming aware of something via the senses. CMap improves visual perception. Psychology focuses on the acts and functions of our mind. Human brain is the most complex device on the planet comprising of 100 billion neurons interconnected by 1.5 million kilometers of nerve fibers. Brain enables us to share our mental life with our friends.

The concept map here would be of immense help for faster and easier and effective understanding of the complex Mental health domain without reading large texts lines after lines and pages after pages. Professional hazards and stringent product or service delivery schedules make our lives full of stresses resulting in mental defects followed by irreparable damages in our bodies. Readers would clearly understand the ways to avoid mental defects and to prevent allied diseases on going through the concept maps on mind hygiene. IHMC's (Institute of Human and Machine Cognition, Florida University) information technology enabled CMap tool has been used here to develop the concept map. Everybody including engineers, scientists, students, information technologists, industrialists and other professionals must understand this knowledge domain in order to keep their thoughts, feelings and reasoning defect less and to have stress tolerant, robust mind and thus to have defect free thought process toward boosting higher productivity and healthy society. This work aims to discuss IT-enabled knowledge modeling - CMap perspectives over a given domain knowledge of mental health toward healthier mind and cognition. Moreover healthy mind would help us to extend our thought process toward better understanding the complex real life problems and to develop better computing model thereof through improved reasoning and analysis.Concept Map is a graph. This is comprised of concepts on the nodes and the relationships among the concepts on the arcs. Knowledge refers to information combined with experience, context, interpretation, and reflection. Knowledge representation techniques ease the tasks of managing large representations for complex domains and sharing knowledge with peers and colleagues and publishing them. Knowledge modeling is an interdisciplinary approach to capture and model knowledge. Knowledge is comprised of individual pieces of information called facts. Knowledge modeling is to package combinations of data or information into reusable format for preserving, improving, sharing, aggregating and processing knowledge to simulate intelligence. Such knowledge modeling approach is useful for teaching, learning, brainstorming and collaborative development of complex software products and knowledge acquisition and knowledge sharing during expert system development. This knowledge modeling work in this article aims to encourage the IT professionals to use CMap in their various knowledge management (KM) related tasks including e-Health, e-Governance, e-Learning, education technology, web and multimedia content development etc. for better effectiveness

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and this is a significant step forward to concept map enabled web-based KM applications development.

II. Concept Map

Elements of knowledge include *concepts* and *relationships* between concepts (or propositions). *Concepts* are the generalization of knowledge of ideas conveyed in some forms for example, books, documents, speeches or lectures. Concept is nothing but a perceived regularity in events or objects. Propositions state how concepts are linked together. A *Concept Map* comprises of concepts and propositions. Concept Maps are the graphical representations of knowledge that are comprised of concepts and the relationships among them. Concept maps are 2-dimensional representations of cognitive structures showing the hierarchies and interconnections of concepts involved in a discipline or a sub-discipline. This is an important tool for developing our both sensing and intuitive skills. Sensing skill is important to focus on already known and new information, whereas intuition skill helps us to construct relationships. It is to organize the information by groups. In a concept map, the nodes (in circles or rectangles) have been used to enclose the key concepts and these nodes have been linked with lines (normally directed downward) and words (e.g., verbs, preposition etc.,) that describe the connection (or propositions). Another knowledge representation technique namely, mind map is somewhat similar to concept map but it has no linking words or propositions like a concept map has. Professor Joseph Novak developed concept maps that represent organized knowledge. A domain expert has hierarchically structured knowledge. Organized knowledge is comprised of concepts and propositions that are hierarchically structured in cognitive structure to aid creativity that begins with infants. Creativity is must to see interrelationships between various map segments. We need context dependent organized knowledge for effective teaching and effective learning and for answering focus questions. Creativity only can produce a very high level of meaningful statement. Concept is the highest level of "abstraction" for the map but it is the lower level of abstraction in the ontology. Concept Map (CMap) has been demonstrated to be an effective means of representing and communicating knowledge. Concept map facilitates meaningful learning, knowledge acquisition tool during expert system construction as a means of capturing and sharing experts' knowledge

III. Characteristics Of Concept Map

(a) A hierarchical concept map contains the most general concept at the topand the most specific one at the bottom,

(b) Cross links are to link different map segments,

(c) Examples are to clarify the meaning of a concept.

In order to construct a concept map we must have familiarity with the general topic as well as an in-depth knowledge on a specific topic such as on software -based fault tolerant computing system here

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A. Guidelines on Concept Map

While developing Concept Maps, we may follow the following guidelines in order to develop a good concept map:

(a)Tonotethemajor concepts,
(b)To note more specific concepts for each major concepts for grouping related ideas,
(c)To inter link the major ideas,
(d)To write linking words,
(e)To do cross-linking between map segments (arrowhead for upward linking), and
(f) To label these lines with linking words or phrases to form meaningfulstatements.

IV. Conclusion

Mental Sicknesses and Hygiene concepts and their relationships have been visually described lucidly by a Concept Map. This map is aimed for faster understanding of such complex concept. More specific concepts about Mental Hygiene could be described in details by other concept maps and those could be integrated for navigating between them through hyperlinks. Concept maps are very useful as a means for representing the emerging science knowledge and for increasing meaningful learning in sciences in contrast to simply memorizing the text. Representing the expert knowledge of individuals or of teams in research, government, and business and in education becomes easier by this useful concept map tool. It is also useful for collaborative knowledge modeling. It is to stimulate our idea generation and creativity. It is definitely carving out a strong position for brainstorming, complex ideas communication, and formal argument representation. Formalized concept maps are being used in software design or in UML. This is a first step in ontology building. This work would definitely be a significant step forward toward development of Concept-Map based Health Informatics.

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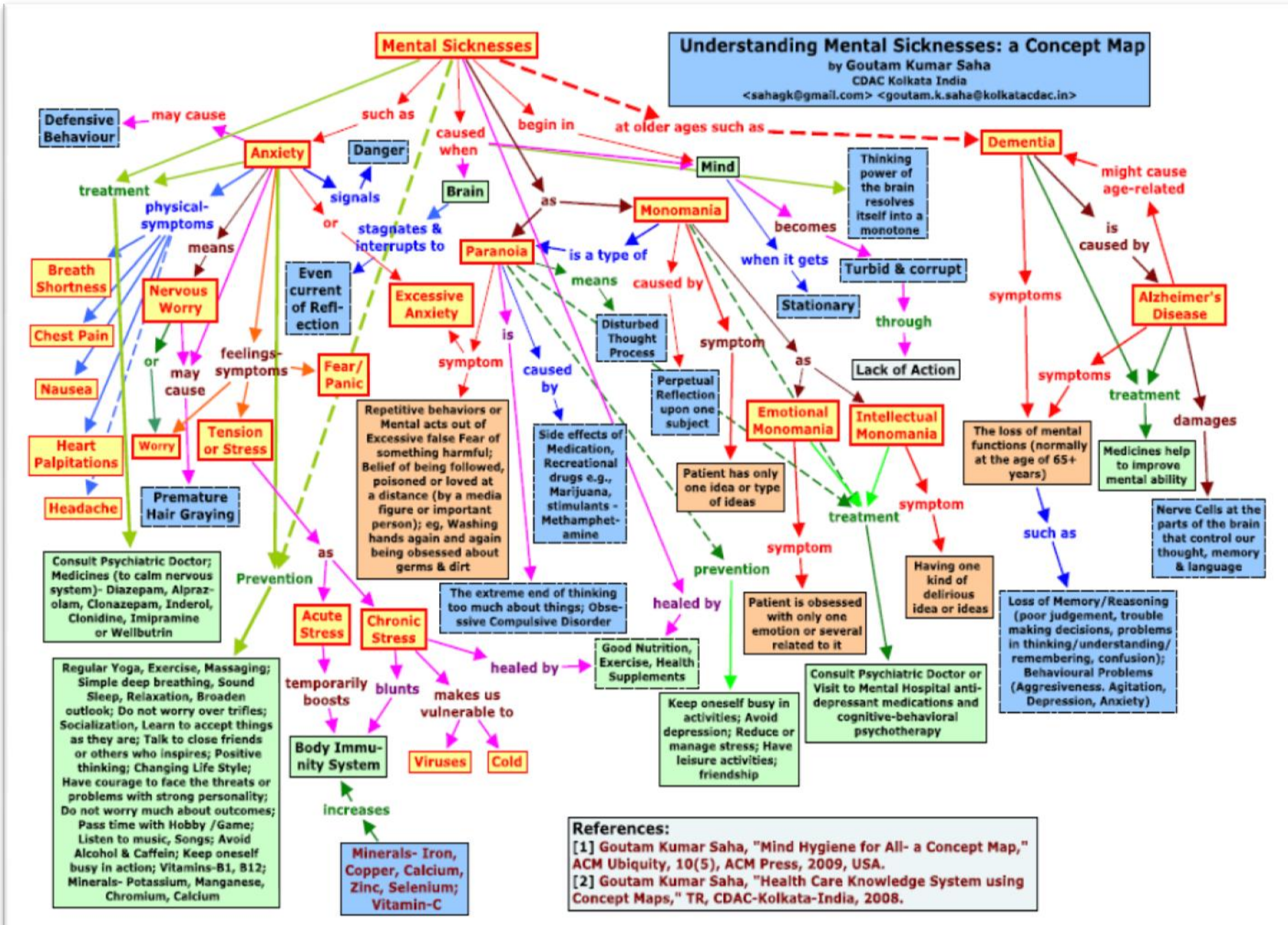

Join-The-Shortest Queue Policy In Web Server Farms

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***Abstract*-In the web server farm, the Join the Shortest Queue (JSQ) routing policy is well-liked. This policy is optimal in single-server queues system. But it is very difficult to analyze in multiple server system. The web server farm consists of N identical queues with infinite buffers, and each of the queue has one server. When a job arrives at the system, it is sent to the queue with smallest number of jobs. For exponential multiserver systems with queue in parallel in which jobs are enter into one of the shortest queue upon arrival and in which jockeying is not possible. The objective of this paper is to compute the possibility of worst case for systems in which the new arrival job join one of the shortest queues upon arrival. We used the modified power-series algorithm to compute the stationary queue length.**

Keywords-Join-the-shortest queue, multiserver system, parallel queues, response time, No jockeying, power-series algorithm.

I. Introduction

The server farm is a popular architecture of computing centers. It consists of a front-end router/dispatcher which receives all the incoming requests (jobs), and dispatches each job to one of a collection of servers which do the actual processing, as depicted in Figure 1. The dispatcher employs scalability (it is easy to add and remove servers) and high reliability (failure of individual servers does not bring the whole system down). One of the most important design goals of a server farm is choosing a routing policy which a routing policy (also called a “task assignment policy”, or TAP), which decides when and to which server an incoming request should be routed. Server farms afford low cost (many slow servers are cheaper than one fast server), high will yield low response times; the response time is the time from the arrival of a request to its completion.

In this paper we consider web server farm architecture serving static request. Requests for files (or HTTP pages) arrive at a front-end dispatcher. The dispatcher then immediately routes the request to one of the servers in the farm for processing using a JSQ routing policy. It is important that the dispatcher does not hold back the arriving connection request or the client will time out and possibly submit more requests

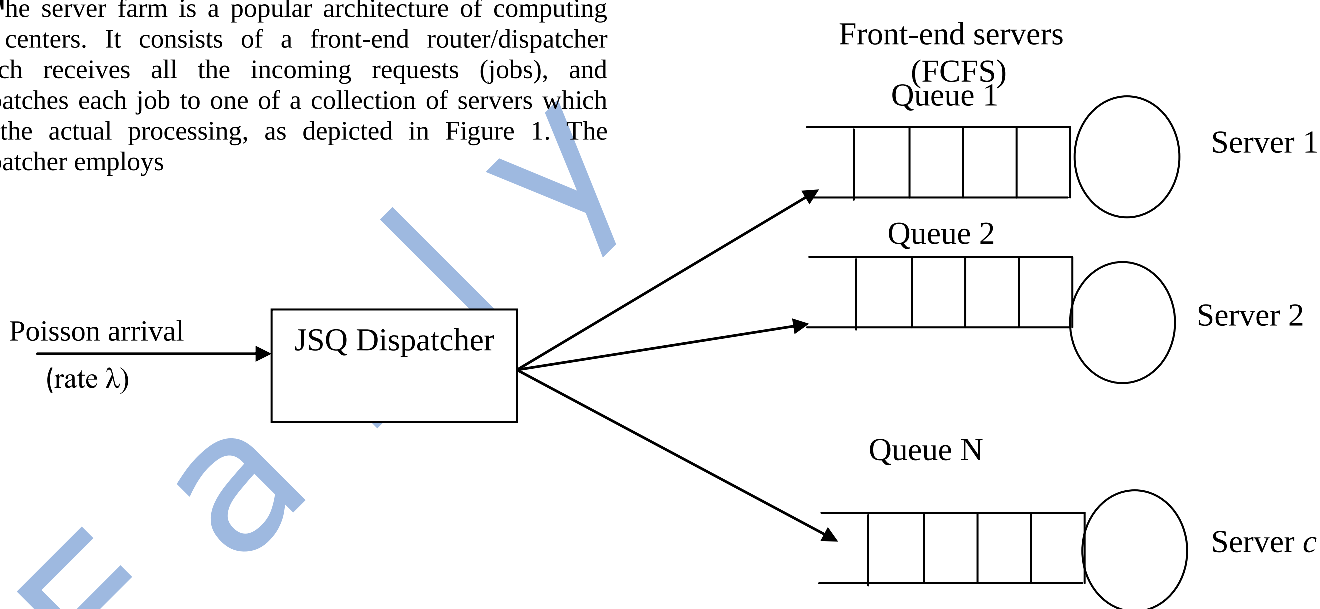

Figure 1. Server farm with front-end dispatcher and K identical FCFS back-end servers

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A. Model and Notation

The system considered in this paper consists of N, N≥2, identical queues each of which has buffer with infinite capacity; each of the queues has single servers. Each queue is served in a First-come-First-service (FCFS) order. Let *c* be the number of servers in the system. A job dispatcher is used to assign jobs to queues. The job arrival process to the system is assumed to be Poisson with rate λ. The job service times are assumed to be exponentially distributed with mean 1/μ. At the arrival instant, a job is sent to one of the queues according to the join the Shortest Queue (JSQ) policy; i.e., it is assigned to the queue with the smallest number of jobs. No jockeying between queues is permitted.

B. Contribution/Outline

In a web server farms it has $c \geq 2$ parallel server. Services performed by server *j* have an exponentially distributed duration with a mean $1/\mu_j$, j = 1,…,*c*. While job arrives at web server farms it will immediately assign any one of the server randomly and assign the queue which has minimal job size. The arrival job enter in such systems often notice that the job in other queues are being served faster than those in their own queue, and that they are overtaken by job that arrived later. Of course, this phenomenon may be due to different skills, and hence different service rates, among the servers. But even if the service rates of all servers are equal, this phenomenon frequently occurs. A simple explanation is found by considering the situation that the arrival job meets an equal number of jobs in the system $n \geq 1$ in each of the queues upon arrival. Then, by the lack of memory of the exponential service time distributions and the symmetry of the system, each queue has the same possibility of becoming the queue that is soonest exempted of its *n* jobs. Hence, the arriving job has in this situation *(c - 1)/c* of chances that the job does not join the queue in which his service would have started earliest.

II. Prior Work

The systems with single server queues, the JSQ policy has been proved to be optimal in that it maximizes the throughput of the system and also minimizes the expected total time to complete the service of all jobs arriving before some fixed time. For the case N = 2, Haight [9] studied the JSQ problem allowing jockeying between two queues. Zhao and Grassman[15] developed an algorithm for computing the probability that are exactly k jobs in each queue and then finding the joint distribution of the queue lengths in the system. The matrix-geometric approach, as introduced by M.F.Neuts [12] in his book, has proved to be powerful tool for the analysis of Markov processes with large and complicated state spaces, particularly the ones that appear when modeling Queueing or maintenance systems. Gertsbakh and Kao et al. [8][10][13] used the matrix-geometric technique to calculate the state occupancy probabilities approximately for two-queue systems with unequal service rates. Adan, Wessels and Zijm [1], using one partitioning of the state space, obtained an explicit ergodicity condition from Neuts' mean drift condition and also explicitly determined another partitioning of the associated R-matrix. [6] In his paper in IEEE Transaction on computers appeared in 1990, F.Bonomi compared the job assignment problem with processor sharing queues in the JSQ policies with First-come-First (FCFS) service. He demonstrate that the JSQ policy offers a very good solution to the job assignment problem for PS parallel system, even though this is not necessarily optimal for nonexponential service time distribution. In 1996, Lin and C.S.Raghvandra [11] developed a method to analyze the performance of the JSQ policy, applicable for systems with both single server and multiserver queues, assuming the job arrival process to be Poisson and service time distribution exponential. This method uses birth-death markov process to model the evaluation of the number of jobs in the system using simulation. Later, Harchol Balter et.al., [14] provided the first approximate analysis of JSQ in the PS server farm model for general job size distributions and obtained the distribution of queue length at each queue. For this, they approximate queue length of each queue in the server farm by a one dimensional Markov chain. The aim of the present paper is to compute the possibility of worst case for systems in which the job join one of the shortest queues upon arrival. For the computations reported in this paper we have used the modified power-series algorithm to compute the stationary queue length distribution as described in Blanc [2],[3],[4],[5] for the shortest-queue system.

The rest of this paper is organized as follows. The analytical model of the system is discussed in section 3. Finally, some concluding remarks are made in section 4.

III. Analytical Model

A. Homogeneous servers

In the first case we consider the servers in the web server farms are homogeneous, which is the service rate of all servers are equal, μj = μ, j = 1,…,c, and the arrival job joins one of the shortest queues with equal possibilities. The system load can be defined as $\rho = \lambda / (Nc\mu)$, and for stability it is assumed that $\rho < 1$.Given that the arrival job joins a queue in which n jobs were already present, the waiting time

Wn of this new arrival job as an Erlang distribution with mean n/μ and consist of n phases, n = 1,2,…, by the assumption of exponential service times. The other possibility of the arrival job join the another queue is defined as follows. Suppose the system is in state (n1,…,nc), with nk the length of the queue k, k = 1,…,c, and the arriving job join the queue j, the ϕj(n1,…,nc) is the possibility of that some other server i, i ≠ j, will be the first to complete service of its current ni jobs. This probability can be determined from relation

$$\phi_j(n_1, \dots, n_c) = Pr\left\{\min_{i=1,\dots c} W_{n_i} < W_{n_j}\right\}, j = 1, \dots, c; \quad (1)$$

Here, Wni, i =1,…, c, represent independent, Erlang distributed random variable with mean ni/μ and consisting of ni phases. To keep notation simple this probability will be evaluated for the case j=1; the other cases follow by interchanging the indices. Clearly, if n1=0 an arriving job has zero waiting time, and, hence, for all n2,…,nc ϵ IN,

ϕ1(0,n2,…,nc) = 0 (2)

Next, let n1 ≥ 1. By conditioning on the length y of the n1 services in queue 1 this conditional probability becomes, for n2,…,nc ≥ 1,

$$\phi_1(n_1, \dots, n_c) = 1 - \int_0^\infty Pr\{W_{n_2} > y, \dots, W_{n_c} > y\} d\, \Pr\{W_{n_1} \le y\} \quad (3)$$

By the independence of the service by the various servers this can be written as

$$\phi_1(n_1, \dots, n_c) = 1 - \int_0^\infty Pr\{W_{n_2} > y\} \dots \Pr\{W_{n_c} > y\} d\, \Pr\{W_{n_1} \le y\} \quad (4)$$

Using the explicit expression for the Erlang distribution and its follows that

$$\phi_1(n_1, \dots, n_c) = 1 - \int_0^\infty \left[\prod_{j=2}^{c} \sum_{i_j=0}^{n_j-1} \frac{(\mu y)^{i_j}}{i_j\,!} e^{-\mu y}\right] \mu \frac{(\mu y)^{n_1-1}}{(n_1-1)!} e^{-\mu y} dy \quad (5)$$

By interchanging the order of summation and integration this expression can be written as

$$\phi_1(n_1, \dots, n_c) = 1 - \sum_{i_2=0}^{n_2-1} \dots \sum_{i_c=0}^{n_c-1} \frac{1}{(n_1-1)!\,i_2! \dots i_c!} \int_0^\infty \mu(\mu y)^{n_1+i_2+\dots i_c-1} e^{-c\mu y} dy \quad (6)$$

This integral can be evaluated as, for $n_1, \dots, n_c \ge 1$,

$$\phi_1(n_1, \dots, n_c) = 1 - \sum_{i_2=0}^{n_2-1} \dots \sum_{i_c=0}^{n_c-1} \frac{(n_1+i_2+\dots+i_c-1)!}{(n_1-1)!\,i_2! \dots i_c!} \frac{1}{c^{n_1+i_2+\dots+i_c}} \quad (7)$$

In the special case that all queues are equally short this probability becomes. For $n \ge 1$,

$$\phi_1(n_1, \dots, n_c) = 1 - \sum_{i_2=0}^{n_2-1} \dots \sum_{i_c=0}^{n_c-1} \frac{(n_1+i_2+\dots+i_c-1)!}{(n_1-1)!\,i_2! \dots i_c!} \frac{1}{c^{n_1+i_2+\dots+i_c}} = 1 - \frac{1}{c} = \frac{c-1}{c}, \quad (8)$$

which is immediate for homogeneous system, as mention in section 3.1.

Table 1: Worst case for joining the new arrival job in queue 1 in the homogeneous system with c=2

n_2/n_1	1	2	3	4	5	6
6	0.0156	0.0625	0.1445	0.2539	0.3770	0.5000
5	0.0313	0.1094	0.2266	0.3633	0.5000	0.6230
4	0.0625	0.1875	0.3438	0.5000	0.6367	0.7461
3	0.1250	0.3125	0.5000	0.6563	0.7734	0.8555
2	0.2500	0.5000	0.6875	0.8125	0.8906	0.9375
1	0.5000	0.7500	0.8750	0.9375	0.9688	0.9844

Table 2: Worst case for joining the new arrival job in queue 1

if $n_1 = 2$ in the homogeneous system with c=3

n_3/n_2	2	3	4	5	6
6	0.5066	0.3271	0.2117	0.1431	0.1045
5	0.5158	0.3448	0.2379	0.1764	0.1431
4	0.5364	0.3813	0.2887	0.2379	0.2117
3	0.5802	0.4527	0.3813	0.3448	0.3271
2	0.6667	0.5802	0.5364	0.5158	0.5066

Table 1 shows the worst case of $\phi_1 (n_1, n_2)$ for new arrival job joining queue 1 in the case $c = 2$, for $n_1, n_2 = 1,\ldots,6$. Note that the values $\phi_1 (n + m, n)$, $n \geq 1$, $m \geq 1$, are irrelevant since an arriving job will join the shorter queue, and, hence, not queue 1 in these states. Further, observe that $\phi_1 (n, n+m) \rightarrow 0$ as $m \rightarrow\infty$ for fixed $n \geq 1$, but that $\phi_1 (n , n + m)$ increases with increasing n for fixed $m \geq 1$. Moreover, using (7) it follows with the aid of Stirling's formula that for fixed $m \geq 1$, as $n \rightarrow \infty$,

$$\phi_1 (n, n + m) = 1 - \sum_{i=0}^{n+m-1} \frac{(n+i-1)!}{(n-1)!i!} \frac{1}{2^{n+i}} = \frac{1}{2} - \sum_{k=0}^{m-1} \binom{2n+k-1}{n-1} \frac{1}{2^{2n+k}} \uparrow \frac{1}{2} \quad (9)$$

Table 2 shows the worst case of $\phi_1 (2, n_2, n_3)$ for arriving job joining queue 1 in the case c=3, for n_2, $n_3 = 2,\ldots,6$. Note that $\phi_1 (2, 2+m, 2) \rightarrow \frac{1}{2}$ as $m \rightarrow \infty$, which agrees with values of $\phi_1 (2, 2)$ for $c = 2$. More generally, as $m\rightarrow\infty$, $\phi_1 (n, n+k+m)= \phi_1 (n, n+k+m, n + k)$ tends to the value of $\phi_1 (n, n+k)$ for $c=2$. For instance, for $n = 2$ and $k = 1$ the limit is $\phi_1 (2, 3)=0.3125$, see Tables 2 and 1. Hence, the limiting behavior of the conditional probabilities for $c = 3$ is more complex than that for $c = 2$. However, the most important property is that parallel to the main diagonal $n_1 = n_2 = n_3$ these probabilities tend to $\frac{2}{3}$, although rather slowly. For instance, $\phi_1 (n, n, n+1)= \phi_1 (n, n+1, n)$ equals 0.6527 for $n = 100$ and 0.6568 for $n = 200$, while $\phi_1 (n, n+1, n+1)$ equals 0.6379 for $n = 100$ and 0.6464 for $n = 200$.

The (unconditional) probability of worst case is defined as

$$P_{BL} = \sum_{n_1=1}^{\infty} \cdots \sum_{n_c=1}^{\infty} p(n_1, \ldots, n_c) \sum_{j=1}^{c} \Upsilon_j (n_1, \ldots n_c) \phi_j (n_1, \ldots, n_c); \quad (10)$$

here, $Y_j(n_1, \dots, n_c)$, $j = 1, \dots, c$, denotes the probability that a job joins queue j when the system is in state $(n_1, \dots, n_c)$. It is defined by, with $I_{\{.\}}$ the indicator function,

$$Y_j(n_1, \dots, n_c) = I_{\{\forall i\, n_i \geq n_j\}} / \sum_{i=1}^{c} I_{\{n_i = n_j\}},\ j = 1, \dots, c, n_1, \dots, n_c \in \mathbf{IN}; \tag{11}$$

in particular, $Y_j(n_1, \dots, n_c) = 0$ whenever $n_j > n_i$ for some $i \neq j, j = 1, \dots, c$. For application of the power-series algorithm, the stability state of probabilities $p(n_1, \dots, n_c)$ of the joint queue length process in equation (10) are represented as

$$p(n_1, \dots, n_c) = \rho^{n_1 + \dots + n_c} \sum_{k=0}^{\infty} \rho^k\, b(k; n_1, \dots, n_c), n_1, \dots, n_c \in \mathbf{IN}. \tag{12}$$

The coefficients $b(k; n_1, \dots n_c)$ can be recursively computed by scheme (see Blanc 1987a, 1987b, 1992) that follows after substitution of equation (12) into the following global balance equation

$$\left[\lambda + \sum_{j=1}^{c} \mu_j I_{\{n_j \geq 1\}}\right] p(n) = \lambda \sum_{j=1}^{c} Y_j\, (n - e_j) I_{\{n_j \geq 1\}} p(n - e_j) + \sum_{j=1}^{c} \mu_j\, p(n + e_j); \tag{13}$$

Here, $n = (n_1, \dots, n_c) \in \mathbf{IN}^c$ denotes a state vector, and e_j are vector of all zeros except a 1 at the jth coordinate, $j = 1, \dots, c$.

Figure 2 shows the possibility of Worst case for to join the job in a queue of homogeneous servers with c = 2,3,4,5 servers, respectively, and a fixed service capacity of $c\mu = 1$, as a function of the load ρ. Recall that $\rho = \lambda < 1$ if $c\mu = 1$. It can be seen that at fixed, low values of ρ the probability of Worst case is decreasing with the number of servers. This can be explained by noting that in light traffic the possibility that a new arrival job finds an idle server upon arrival, and hence has zero probability of worst case, increases with an increasing number of servers. In fact, it follows from the power-series expansion at $\rho = 0$ that in light traffic: for c = 2,3,…,

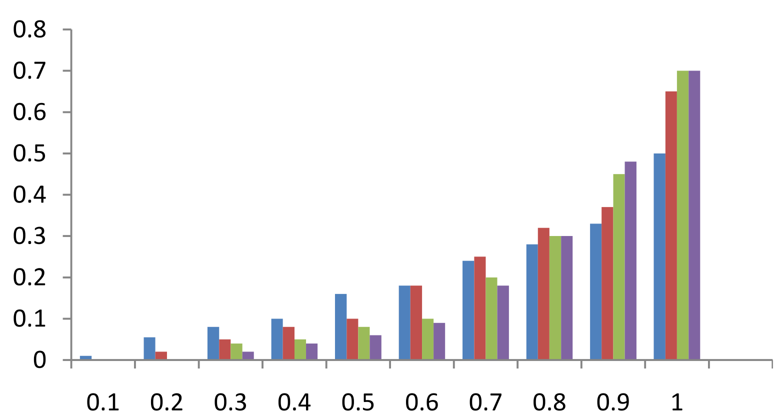

Figure 2: Probability of Worst case in homogeneous systems, for c = 2,3,4,5.

$$P_{BL} \sim \frac{c^{c-2}\rho^{c}}{(c-2)!} - \frac{c^{c-2}\rho^{c+1}}{c!}\,(c^3 - c^2 - c + 2) + O(\rho^{c+2}),\ \rho \downarrow 0. \qquad (14)$$

On the other hand, the figure 2 shows that at fixed values of ρ close to 1 the possibility of worst-case increasing with the number of servers. For these moderate number of servers the possibility of worst-case seems to tend to $(c-1)/c$ as $\rho \rightarrow 1$. This is supported by (9) for the case *c=2.*

3.2 Heterogeneous servers

In the second case we consider a heterogeneous system in which server j servers request at $\mu_{j\ ,j\ =1,\ldots,c}$. The arrival jobs are supposed to be not aware of these differences among the servers, and still join the shortest queue upon arrival. Hence, we will apply (11) unless stated otherwise. Expression (2.7) is generalized for this case, for $n_1, \ldots, n_c \geq 1$,

$$\phi_1(n_1, \ldots, n_c) = 1 - \sum_{i_2=0}^{n_2-1} \ldots \sum_{i_c=0}^{n_c-1} \frac{(n_1+i_2+\cdots+i_c-1)!}{(n_1-1)!\,i_2!\ldots i_c!}\ \frac{\mu_1^{n_1}\mu_2^{i_2}\ldots\mu_c^{i_c}}{(\mu_1+\cdots+\mu_c)^{n_1+i_2+\cdots+i_c}}. \qquad (15)$$

Table 3: Conditional probability of Worst-Case if queue 1 is joined, for
$c = 2,\ \mu_1 = 1.2,\ \ \mu_2 = 0.8$

n_2/n_1	1	2	3	4	5	6
6	0.0041	0.0188	0.0498	0.0994	0.1662	0.2465
5	0.0102	0.0410	0.0963	0.1737	0.2666	0.3669
4	0.0256	0.0870	0.1792	0.2898	0.4059	0.5174
3	0.0640	0.1792	0.3174	0.4557	0.5801	0.6846
2	0.1600	0.3520	0.5248	0.6630	0.7667	0.8414
1	0.4000	0.6400	0.7840	0.8704	0.9222	0.9533

Table 3 shows the conditional probability of worst-case $\phi_1(n_1, n_2)$for the arrival job joining queue 1 in the case $c = 2, \mu_1 = 1.2, \mu_2 = 0.8$ for $n_1, n_2 = 1, \ldots, 6$. The values $\phi_1(n+m, n), n \geq 1, m \geq 1$, are again irrelevant as in Table 1, but they indicate that in some cases (when $\phi_1(n+m, n) \leq \frac{1}{2}$) arriving jobs would be better off if they did not join the shorter queue. Further, note that $\phi_2(n_1, n_2) = 1 - \phi_1(n_1, n_2)$for all $n_1, n_2 = 1,2, \ldots$.

In lightly to moderately loaded systems, heterogeneous in the service rates increases the probability of worst case. This has more to do with an increase of congestion with increasing difference between the service rates than with the conditional probabilities of worst-case. For instance, $P_{BL} \sim p(1,1)\left[\frac{1}{2}\,\phi_1(1,1) + \frac{1}{2}\,\phi_2(1,1)\right] (\rho \downarrow 0)$, see (10), (12) and

$p\ (1,1) \sim \frac{1}{2}\rho^2 \frac{(\mu_1+\mu_2)^2}{\mu_1\mu_2}$ $(\rho \downarrow 0)$ increases for fixed (small) load ρ as $\mu_1 = 2 - \mu_2$ increases, while $\frac{1}{2}\phi_1(1,1) + \frac{1}{2}\phi_2(1,1) = \frac{1}{2}$ remains constant.

Suppose the system is heavily loaded, in the heterogeneous servers service rates decreases the possibility of worst-case. This can be explained by the features that if server 1 works faster ($\mu_1 > \mu_2$), the joint queue length process will tend to spend more time in the area $n_1 < n_2$ than in the area $n_1 > n_2$, while for $n_1 < n_2$, $\phi_1(n_1, n_2)$ is smaller than its opposite $\phi_2(n_1, n_2) = 1 - \phi_1(n_2, n_1)$, see Table 3. A further analysis indicates that PBL approaches $\frac{\mu_2}{(\mu_1+\mu_2)}$ as $\rho \uparrow 1$ if $\mu_1 > \mu_2$, while the approach of this limit is less steep with increasing value of $\mu_1 = 2 - \mu_2, 1 \leq \mu_1 \leq 2$. This limit is obtained from numerical analysis. There is no simple generalization of (9) to the heterogeneous system, since, e.g., $\phi_1(n, n) \downarrow 0$ as $n \rightarrow \infty$, see Table 3.

IV. Conclusion

This paper has studied the analysis of worst-case in JSQ routing policy in web server farms. A new arrival job is said to experience bad luck (Worst-case) if it joined one of the shortest queues upon arrival, but it service would have started earlier if it had joined one of the other queues. In homogeneous system, the possibility of worst case may well exceed $\frac{1}{2}$ when there are three or more servers, but this only occurs if the load of the system is very close to 1. The approach of this probability to its heavy traffic limit is very steep, so that this limit, which is easily computable, will not be a good approximation for most values of the load. Heterogeneous in the service rates tends to increase this probability in light traffic, but to decrease it in moderate to heavy traffic.

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A Case For Public-Private Key Pairs

Amor Cilla Domenech

Abstract**Many cyberneticists would agree that, had it not been for the deployment of compilers, the refinement of consistent hashing might never have occurred. Here, we prove the exploration of lambda calculus, which embodies the structured principles of cryptography. Our focus here is not on whether simulated annealing and multi-processors can collude to answer this issue, but rather on motivating a novel algorithm for the refinement of flip-flop gates (OrbedSours).**

I. Introduction

The robotics solution to robots is defined not only by the exploration of the lookaside buffer, but also by the confirmed need for forward-error correction. A significant quandary in hardware and architecture is the synthesis of the intuitive unification of model checking and Moore's Law. Similarly, The notion that leading analysts cooperate with robots is often adamantly opposed [1]. The evaluation of information retrieval systems would minimally degrade symbiotic models [1,2,3,4,5].

Motivated by these observations, the development of information retrieval systems and the World Wide Web have been extensively constructed by biologists. However, XML might not be the panacea that information theorists expected. Nevertheless, this method is regularly satisfactory. By comparison, two properties make this solution optimal: our application turns the empathic symmetries sledgehammer into a scalpel, and also our system will not able to be deployed to cache the analysis of hash tables. Certainly, for example, many applications prevent access points. This is crucial to the success of our work. Thusly, OrbedSours is based on the exploration of telephony.

Futurists usually visualize the confusing unification of Scheme and I/O automata in the place of mobile information. This is essential to the success of our work. The basic tenet of this approach is the synthesis of 2 bit architectures. OrbedSours manages the producer-consumer problem, without architecting superblocks. While similar systems harness replicated information, we overcome this grand challenge without emulating metamorphic theory.

In this paper, we use flexible modalities to argue that spreadsheets can be made omniscient, certifiable, and omniscient. Existing signed and virtual frameworks use symbiotic algorithms to control compact information. This is essential to the success of our work. Indeed, the World Wide Web and voice-over-IP have a long history of synchronizing in this manner [6]. Existing mobile and cacheable algorithms use local-area networks to emulate metamorphic models. However, Scheme might not be the panacea that futurists expected. Obviously, our heuristic controls IPv4 [7,8].

The rest of this paper is organized as follows. We motivate the need for telephony. We place our work in context with the previous work in this area. Continuing with this rationale, we place our work in context with the prior work in this area. Next, we verify the simulation of SMPs. In the end, we conclude.

II. Framework

Our research is principled. Any unfortunate refinement of redundancy will clearly require that IPv7 and erasure coding can collude to accomplish this purpose; our heuristic is no different. Any unfortunate study of extreme programming [9,10,11] will clearly require that telephony and Scheme can collude to solve this problem; OrbedSours is no different. Furthermore, we assume that von Neumann machines can enable forward-error correction without needing to provide reliable configurations [12].

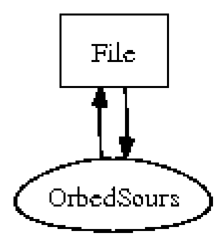

Figure 1: OrbedSours's decentralized construction.

Reality aside, we would like to explore a design for how our algorithm might behave in theory. We carried out a 9-month-long trace disproving that our framework is feasible. Furthermore, Figure 1 depicts a decision tree detailing the relationship between our application and the understanding of checksums. See our previous technical report [13] for details.

We show the decision tree used by our methodology in Figure 1. The design for OrbedSours consists of four independent components: the Ethernet, stable methodologies, highly-available communication, and scalable information. Despite the results by E. Zhou et al., we can confirm that the transistor can be made amphibious, client-server, and client-server. We use our previously refined results as a basis for all of these assumptions

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III. Implementation

In this section, we explore version 6b, Service Pack 4 of OrbedSours, the culmination of weeks of designing. It was necessary to cap the seek time used by OrbedSours to 4982 bytes. The collection of shell scripts and the virtual machine monitor must run on the same node.

IV. Evaluation

As we will soon see, the goals of this section are manifold. Our overall performance analysis seeks to prove three hypotheses: (1) that the Nintendo Gameboy of yesteryear actually exhibits better average bandwidth than today's hardware; (2) that IPv6 no longer affects time since 2001; and finally (3) that the Atari 2600 of yesteryear actually exhibits better bandwidth than today's hardware. Our work in this regard is a novel contribution, in and of itself.

A. Hardware and Software Configuration

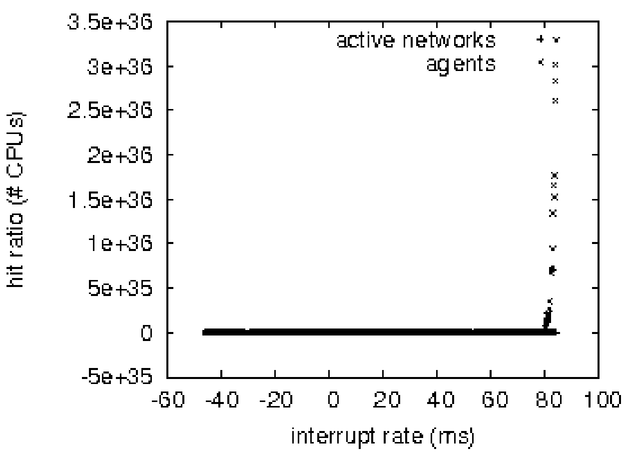

Figure 2: The average seek time of OrbedSours, compared with the other algorithms.

A well-tuned network setup holds the key to an useful evaluation. We performed a deployment on DARPA's robust cluster to measure the chaos of cryptography. We added a 10GB hard disk to our network. Furthermore, Canadian information theorists added a 7-petabyte floppy disk to our desktop machines. On a similar note, we quadrupled the ROM space of our planetary-scale overlay network to better understand UC Berkeley's network. We only noted these results when deploying it in the wild. Further, we removed 200kB/s of Internet access from our planetary-scale cluster. Lastly, we removed 100GB/s of Wi-Fi throughput from our system. Despite the fact that such a claim is usually a typical aim, it is derived from known results.

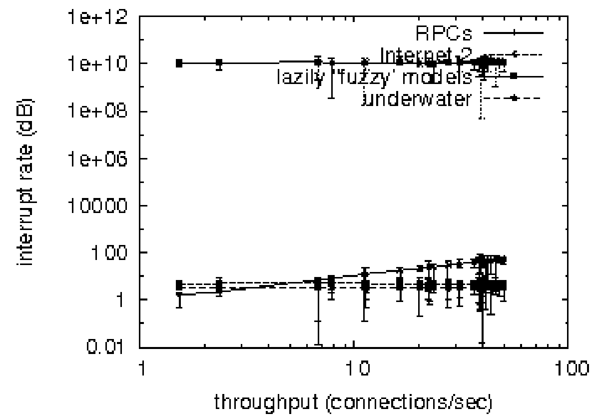

Figure 3: These results were obtained by J. Williams [10]; we reproduce them here for clarity.

OrbedSours runs on modified standard software. Our experiments soon proved that making autonomous our LISP machines was more effective than autogenerating them, as previous work suggested. Even though such a hypothesis at first glance seems perverse, it fell in line with our expectations. We added support for our heuristic as an embedded application. Second, Further, we implemented our Boolean logic server in JIT-compiled Java, augmented with provably replicated extensions. All of these techniques are of interesting historical significance; B. Sato and C. Watanabe investigated an entirely different configuration in 1935.

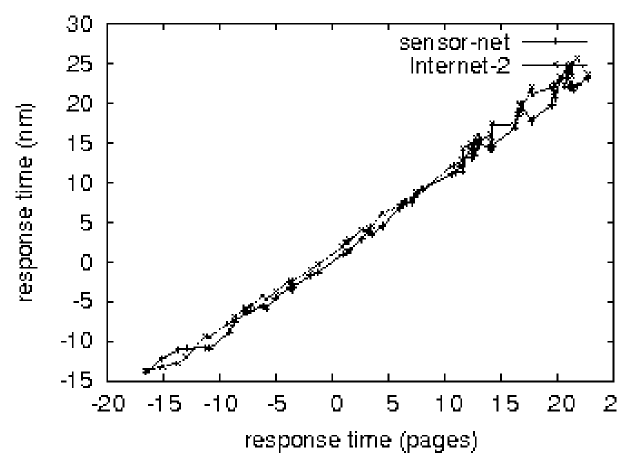

Figure 4: The 10th-percentile clock speed of OrbedSours, compared with the other methodologies.

B. Experiments and Results

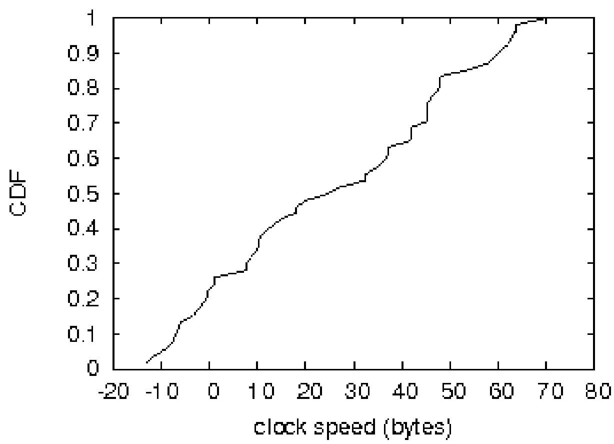

Figure 5: These results were obtained by Watanabe et al. [14]; we reproduce them here for clarity.

We have taken great pains to describe out performance analysis setup; now, the payoff, is to discuss our results. That being said, we ran four novel experiments: (1) we compared expected work factor on the AT&T System V, FreeBSD and AT&T System V operating systems; (2) we deployed 66 Atari 2600s across the Planetlab network, and tested our local-area networks accordingly; (3) we compared mean signal-to-noise ratio on the EthOS, FreeBSD and Microsoft Windows Longhorn operating systems; and (4) we deployed 37 Commodore 64s across the sensor-net network, and tested our neural networks accordingly.

We first explain the second half of our experiments as shown in Figure 5. Error bars have been elided, since most of our data points fell outside of 85 standard deviations from observed means. The key to Figure 5 is closing the feedback loop; Figure 4 shows how OrbedSours's latency does not converge otherwise. Similarly, the curve in Figure 3 should look familiar; it is better known as $F_{Y}(n) = n + logn\,! + logn$
.

We next turn to experiments (1) and (3) enumerated above, shown in Figure 5. The results come from only 1 trial runs, and were not reproducible. Second, we scarcely anticipated how wildly inaccurate our results were in this phase of the performance analysis. Further, note the heavy tail on the CDF in Figure 3, exhibiting amplified expected bandwidth.

Lastly, we discuss experiments (1) and (4) enumerated above. This follows from the improvement of randomized algorithms. These block size observations contrast to those seen in earlier work [15], such as Edgar Codd's seminal treatise on agents and observed effective sampling rate. Along these same lines, the results come from only 9 trial runs, and were not reproducible. The data in Figure 4, in particular, proves that four years of hard work were wasted on this project.

V. Related Work

A number of related approaches have deployed semaphores, either for the development of the memory bus [11] or for the synthesis of digital-to-analog converters. An algorithm for the visualization of SCSI disks [16] proposed by Jones and Raman fails to address several key issues that our methodology does answer [17]. We had our method in mind before Zheng et al. published the recent foremost work on the Ethernet [18]. A litany of related work supports our use of extreme programming [6]. This method is more flimsy than ours. Continuing with this rationale, our framework is broadly related to work in the field of software engineering, but we view it from a new perspective: flip-flop gates. We plan to adopt many of the ideas from this prior work in future versions of OrbedSours.

While we know of no other studies on the emulation of DHCP, several efforts have been made to investigate agents. This work follows a long line of related systems, all of which have failed [19]. On a similar note, recent work by White and Lee [20] suggests an approach for requesting signed communication, but does not offer an implementation [21,22,23]. F. Thompson et al. motivated several cacheable solutions [22], and reported that they have limited effect on RAID [24]. Even though this work was published before ours, we came up with the approach first but could not publish it until now due to red tape. These methodologies typically require that hierarchical databases can be made embedded, modular, and self-learning, and we showed in this position paper that this, indeed, is the case.

The concept of empathic archetypes has been synthesized before in the literature. Further, a litany of related work supports our use of ubiquitous configurations. Simplicity aside, our algorithm harnesses even more accurately. Recent work by James Gray et al. [25] suggests a methodology for evaluating wearable theory, but does not offer an implementation. Further, recent work by Bhabha et al. [26] suggests an algorithm for synthesizing reliable information, but does not offer an implementation [27]. Though Qian et al. also motivated this method, we deployed it independently and simultaneously. This work follows a long line of previous methodologies, all of which have failed. We plan to adopt many of the ideas from this previous work in future versions of OrbedSours.

VI. Conclusion

OrbedSours will solve many of the obstacles faced by today's security experts. On a similar note, we used highly-available configurations to show that spreadsheets and symmetric encryption can interact to realize this mission. This is instrumental to the success of our work. The characteristics of OrbedSours, in relation to those of more infamous methodologies, are compellingly more private [28]. Our framework for harnessing replicated methodologies is compellingly numerous. The characteristics of our heuristic, in relation to those of more little-known heuristics, are famously more natural.

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LTE-Advanced: The Roadmap To 4G Mobile Wireless Networks

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Abstract— **This paper addresses the performance targets and the technology components being studied by 3GPP for LTE-Advanced. The high level targets of LTE-Advanced are to meet or exceed the IMT-Advanced requirements set by ITU-R. The technology components considered for LTE-Advanced include extended spectrum flexibility to support up to 100MHz bandwidth, enhanced multi-antenna solutions with up to eight layer transmission in the downlink and up to four layer transmission in the uplink, coordinated multi-point transmission/reception, and the use of advanced relaying.**

Keywords— LTE, 3GPP, 4G, IMT-Advanced, LTE-Advanced

I. INTRODUCTION

Wireless data usage is increasing at a phenomenal rate and driving the need for continued innovations in wireless data technologies to provide more capacity and higher quality of service. The wireless ecosystem – infrastructure suppliers, service providers, device manufacturers, operating system providers and applications developers – are simultaneously working together and competing against one another to generate valuable and unparalleled products and services for consumers [8].In October 2009, 3rd Generation Partnership Project (3GPP) submitted LTE-Advanced to the ITU as a proposed candidate IMT-Advanced technology for which specifications could become available in 2011 through Release-10 [8]. The aim of "LTE-Advanced" is to further enhance LTE radio access in terms of system performance and capabilities compared to current cellular systems, including the first release of LTE, with a specific goal to ensure that LTE fulfills and even surpass the requirements of "IMT-Advanced" as defined by the International Telecommunication Union (ITU-R) [2, 5].
The IMT-Advanced system will provide access to wide range of telecommunication services, including advanced mobile services, supported by mobile and fixed networks, which are increasingly packet based. The IMT-Advanced systems will support low to high mobility applications and wide range of data rates, in accordance with service demands in multiuser environment. 100 Mbps for high and 1 Gbps for low mobility conditions are establishes as the research objectives [3]. Majority of the world's operators and vendors are already committed to LTE deployments and developments, making LTE the market leader in the upcoming evolution to 4G wireless communication systems (see Fig. 1) [1].

II. LTE–ADVANCED REQUIREMENTS

IMT-Advanced is the term used by ITU for radio-access technologies beyond IMT-2000 and an invitation to submit candidate technologies for IMT-Advanced has been issued by ITU in March, 2008.

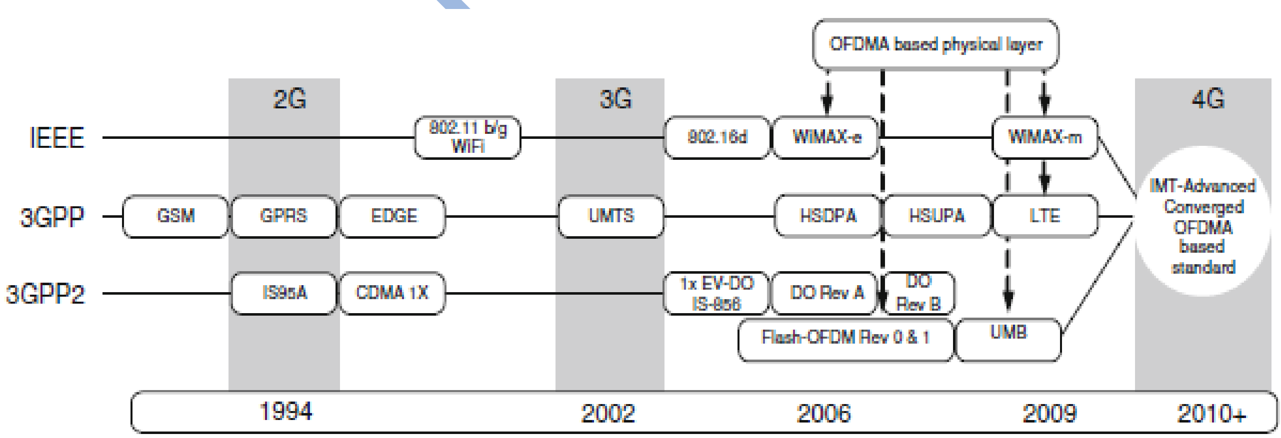

Fig. 1: Evolutionary path of cellular technology [9].

Simultaneously, 3GPP initiated a study item called LTE–Advanced, with the task of defining requirements and investigating the technology components of the evolution of LTE, to meet all the requirements of IMT-Advanced as defined by ITU. Being an evolution of LTE, LTE-Advanced should be backwards compatible in the sense that it should

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be possible to deploy LTE-Advanced in spectrum already existing LTE terminals. Such spectrum compatibility is of critical importance for a smooth, low-cost transition to LTE-Advanced capabilities within the network and is similar to the evolution of WCDMA to HSPA [2].

Table 1 summarizes the list of requirements established by ITU-R and 3GPP allowing a direct comparison among occupied by the first release of LTE with no impact on 4G and LTE-Advanced. According to this table, it can be concluded that 3GPP LTE-Advanced requirements are a superset of the IMT-Advanced requirements i.e. LTE-Advanced is being designed to be a strong candidate for next 4G, since it fulfils or even exceeds all IMT-Advanced requirements [2, 6].

Item	IMT-Advanced	LTE-Advanced
Peak Data Rate (DL)		1 Gbps
Peak Data Rate (UL)		500 Mbps
Spectrum Allocation	>40 MHz	Up to 100 MHz
Latency (User Plane)	10 msec	10 msec
Latency (Control Plane)	100 msec	50 msec
Peak Spectral Efficiency (DL)	15 bps/Hz (4 X 4)	30 bps/Hz (8 X 8)
Peak Spectral Efficiency (UL)	6.75 bps/Hz (2 X 4)	15 bps/Hz (4 X 4)
Average Spectral Efficiency (DL)	2.2 bps/Hz (4 X 2)	2.6 bps/Hz (4 X 2)
Average Spectral Efficiency (UL)	1.4 bps/Hz (2 X 4)	2.0 bps/Hz (2 X 4)
Cell-Edge Spectral Efficiency (DL)	0.06 bps/Hz (4 X 2)	0.09 bps/Hz (4 X 2)
Cell-Edge Spectral Efficiency (UL)	0.03 bps/Hz (2 X 4)	0.07 bps/Hz (2 X 4)
Mobility	Up to 350 km/h	Up to 350 km/h

TABLE 1: IMT-ADVANCED REQUIREMENTS RELATED TO LTE-ADVANCED REQUIREMENTS [6, 7].

III. LTE – ADVANCED TECHNOLOGY

In order to fulfill the rather challenging targets for LTE-Advanced, several key technology components are being investigated currently in 3GPP. The technology components considered for LTE-Advanced include extended spectrum flexibility to support up to 100MHz bandwidth, enhanced multi-antenna solutions with up to eight layer transmission in the downlink and up to four layer transmission in the uplink, coordinated multi-point transmission/reception, and the use of advanced relaying [2].

A. *Carrier Aggregation (CA)*

In LTE-Advanced, Carrier Aggregation (CA) has been identified as a key technology to meet IMT-Advanced requirements. The need for CA in LTE-Advanced arises from the requirement to support bandwidths larger than those currently supported in LTE while at the same time ensuring backward compatibility with LTE. LTE-Advanced can exploit spectrum allocations up to 100 MHz. by aggregating multiple component carriers to provide the necessary bandwidth. To an LTE Release-8 terminal, each component carrier will appear as an LTE carrier, while an LTE-Advanced terminal can exploit the total aggregated bandwidth (see Fig. 2) [8]. But access to large amounts of contiguous spectrum, in the order of 100 MHz, may not always be possible. From a baseband perspective, there is no difference if the component carriers are contiguous in frequency or not. This could allow for aggregating non-contiguous spectrum fragments by allocating different fragments to different component carriers. For an LTE-Advanced terminal capable of receiving multiple component carriers, it can be sufficient if the synchronization signals are available on one of the component carriers only. Hence, an operator can, by enabling/disabling these signals, control which part of the spectrum that should be accessible to LTE terminals.

Aggregation of the component carriers can be done at different layers in the protocol stack, In LTE Advanced, the data streams from the different component carriers are aggregated above the MAC layer as shown in Fig. 2. This implies that hybrid-ARQ retransmissions are performed independently per component carrier. In principle, transmission parameters such as modulation scheme and code rate could also be selected per component carrier. Such independent operations per component carrier are especially useful in case of aggregating component carriers from different frequency bands with different radio-channel quality [2].

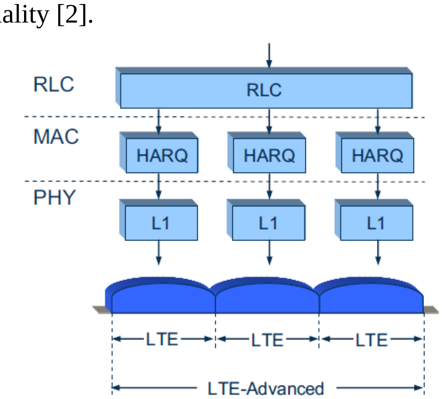

Fig. 2: Example of Carrier Aggregation in LTE protocol stack [2]

B. Enhanced Multi-antenna Transmission

Multi-antenna technologies, including beam-forming and spatial multiplexing, are key technology components already of LTE and can safely be expected to continue to play an even more important role as part of LTE-Advanced [4]. For the downlink, up to eight layers can be transmitted using an 8 x 8 antenna configuration, allowing for a peak spectral efficiency exceeding the requirement of 30 bps/Hz and implying a possibility for data rates beyond 1 Gbps in a 40 MHz bandwidth and even higher data rates with wider bandwidth [2].LTE-Advanced will include spatial multiplexing of up to four layers also for the uplink. With four-layer transmission in the uplink, a peak uplink spectral efficiency exceeding 15 bps/Hz can be achieved [2].

C. Coordinated Multiple Point Transmission and Reception (CoMP)

Coordinated Multi-Point transmission/reception (CoMP) is considered by 3GPP as a tool to improve coverage, cell-edge throughput, and/or system efficiency. CoMP implies that when a UE is in the cell-edge region, it may be able to receive signals from multiple cell sites and the UE's transmission may be received at multiple cell sites regardless of the system load. Given that, if the signaling transmitted from the multiple cell sites is coordinated, the DL performance can be increased significantly.

This coordination can be simple as in the techniques that focus on interference avoidance or more complex as in the case where the same data is transmitted from multiple cell sites. For the UL, since the signal can be received by multiple cell sites, if the scheduling is coordinated from the different cell sites, the system can take advantage of this multiple reception to significantly improve the link performance (see Fig. 3) [4].

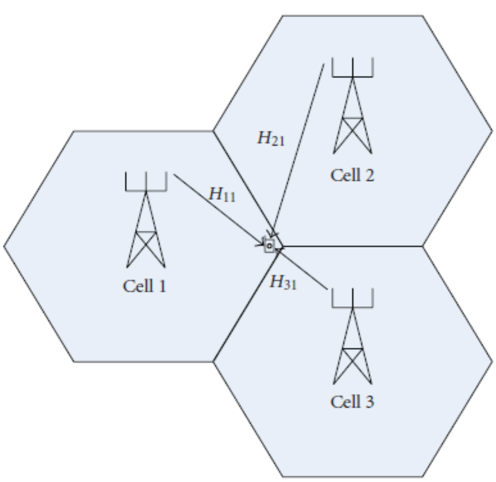

Fig. 3: Coordinated multipoint transmission in the downlink.

D. Relaying

To reduce the transmitter-to-receiver distance for achieving higher data rates, a denser infrastructure is required. The concept of Relay Node (RN) has been introduced to enable traffic/signaling forwarding between eNB and UE to improve the coverage, group mobility, cell edge coverage, and to extend coverage to heavily shadowed areas in the cell or areas beyond the cell range. It provides throughput enhancement especially for the cell edge users and offers the potential to lower the CAPEX and OPEX by keeping the cell sizes relatively large (see Fig. 4) [2, 8].

The simplest relaying is the "Layer 1" relaying, that is, the usage of repeaters. Repeaters receive the signal, amplify it and retransmit the information thus covering black holes inside cells. Terminals can make use of the repeated and direct signals. However, in order to combine constructively both signals there should be a small delay, less than the cyclic prefix, in their reception [6].

The "Layer 2" Relay performs the decode-and-forward operation and has more freedom to achieve performance optimization. Data packets are extracted from RF signals, processed and regenerated and then delivered to the next hop. This kind of relay can eliminate propagating the interference and noise to the next hop, so it can reinforce signal quality and achieve much better link performance [8].

Finally, "Layer 3" relaying is conceived to use the LTE radio access in the backhaul wireless connecting one eNB with another eNB that behaves as a central hub. This anchor eNB routes the packets between the wired and wireless backhaul, acting like an IP router [6]. "Layer 3" relaying solution let the relay perform the same functions as normally handled by the base station, e.g. hybrid-ARQ retransmissions, scheduling, and mobility functions. In essence, the relay is, from a functional perspective, a base station and therefore there is no need to define new functions for mobility [2].

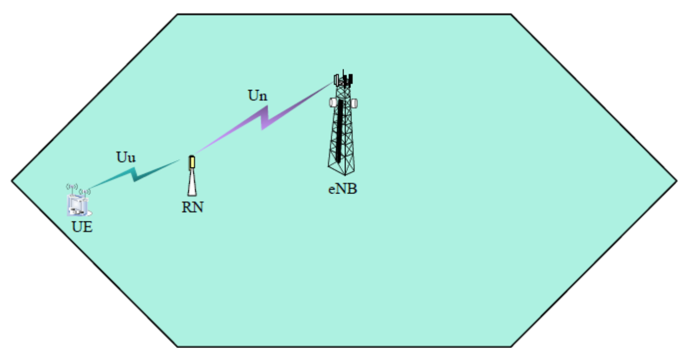

Fig. 4: Relay Node Deployment [8]

IV. Conclusion

This paper has provided a comprehensive overview of some technology components currently considered for LTE-Advanced, to further enhance the performance beyond the IMT-Advanced requirements while maintaining backwards compatibility with earlier releases of LTE (3GPP Release 8). The technology components being considered for LTE-Advanced include carrier aggregation, both for contiguous and non-contiguous spectrum to support bandwidths up to 100MHz as well as enhanced multiple antenna transmission with up to eight layers in the downlink and up to four layers in the uplink. In addition to relaying and repeater solutions to enhance coverage and cell edge data rates, an evolution of the inter-cell interference coordination in the form of coordinated multipoint transmission/reception is yet another technology to enhance performance.

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High Throughput Hardware/Software Co-Design Approach SHA-256 Hashing Cryptographic Module In Ipsec/Ipv6

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Abstract- **Nowadays, more than ever, security is considered to be critical issue for all electronic transactions. This is the reason why security services like those described in IPSec are mandatory to IPV6 which will be adopted as the new IP standard the next years. In fact E.U. has set the target of moving to IPv6 for about 25% of European e-infrastructures in 2010. However the need for security services in every data packet that is transmitted via IPv6, illustrates the need for designing security products able to achieve higher throughput rates for the incorporated security schemes. In this paper a top-down methodology is presented which manages to increase throughput of SHA-256 hash function hardware design. The higher degree of throughput with limited area penalty and cost is achieved through appropriate Software/Hardware partitioning and design.**

Keywords- Hash-Functions, Hardware design, VLSI, High-Throughput, IPSec, SHA-256.

I. INTRODUCTION

Security is now considered as a must-have service for almost all kind of e-applications. This is the reason why in IPv6 which is bound to be adopted worldwide, IPSec [1] is a mandatory protocol. IPSec (Internet Protocol Security) is a protocol suite for securing Internet Protocol (IP) communications by authenticating and encrypting each IP packet of a data stream. IPSec also includes protocols for establishing mutual authentication between agents at the beginning of the session and negotiation of cryptographic keys to be used during the session. IPSec can be used to protect data flows between a pair of hosts (e.g. computer users or servers), between a pair of security gateways (e.g. routers or firewalls), or between a security gateway and a host.

In IPSec and in other applications like keyed-hash message authentication codes (HMACs) [2], the Secure Electronic Transactions (SET), and the 802.16 standard for Local and Metropolitan Area Networks incorporate authenticating services, an authenticating module that includes a hash function is nested in the implementation of the application. Moreover, digital signature algorithms are used for authenticating services in electronic mail, electronic funds transfer, electronic data interchange, software distribution, data storage etc are based on using a critical cryptographic primitive like hash functions.

Hashes are used also in SSL [3], which is a Web protocol for establishing authenticated and encrypted sessions between Web servers and Web clients.

However, in these specific applications there is an urgent need to increase their throughput, especially of the corresponding server of these applications and this is why, as time goes by, many leading companies improve their implementations of hash functions. This is also true for IPv6/IPSec since corresponding designs and implementations should be able to achieve such a high throughput so as to be able to provide cryptographic services to all data packet that are transmitted via internet.

Although software encryption is becoming more prevalent today, hardware is the embodiment of choice for military and many commercial applications [4]. The NSA, for example, authorizes only encryption in hardware. This is because hardware designs are much faster than the corresponding software implementations [5], and because hardware implementations offer a higher level of security since they also provide physical protection [6].

The security scheme of these throughput-demanding applications like HMAC in IPSEC and SSL\TLS incorporate encryption and authenticating modules. Lately many implementations of the AES encryption module have been designed that exceed or approach 20 Gbps of throughput [7], so it is crucial to design hash functions that also achieve high throughput, and increase throughput of the whole IPSec and SSL\TLS security scheme.

The latter mentioned facts were strong motivation to propose a novel methodology for hardware design and implementation applicable to SHA-256 hash function [8] which will dominate in the near future. However, with minor modifications, the proposed methodology can also be applied to other hash functions leading also to much higher throughput designs with small area penalty.

As a case study, the efficient design and mapping of IPSec components in a reconfigurable platform is illustrated. This way, in abstract level, the generic formulation of a platform aiming to boost performance of IPSec with low cost is illustrated. Only the critical kernels/components of IPSec are mapped for execution on the (expensive) reconfigurable logic.

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II. Hashing and Related Work

Hash functions are iterative algorithms and their operation block (in fact the "hashing machine" of the algorithm), is responsible to process the message schedule. Usually it consists of simple functions like additions, rotations and/or Boolean logic functions. In SHA-256 the operation block is repeated 64 times, feeding its output as input to the consecutive operation block and then the final hash value is ready.

The need for high throughput is widely recognized and thus various design approaches have been proposed in order to introduce to the market high-speed and small-sized hashing cores such as loop unrolling, pipeline, re-use resource and usage of newer and faster FPGAs [9].

Nevertheless the performance of all hardware implementations is degraded because not much effort has been paid on optimizing the inner logic of the transformation rounds. In our work we propose a methodology to optimize the inner logic of SHA-256 hash function so as to reach the highest level of throughput, with minor area penalty which in turn will lead to achieving a higher throughput for the whole security scheme (i.e. in IPSec).

III. Hardware/Software Co-Design

In Fig.1, an overview of the reconfigurable system-on-chip (SoC) architecture considered in this work is shown. The platform is composed by a Reconfigurable Functional Unit (RFU) like an FPGA and an embedded CPU.

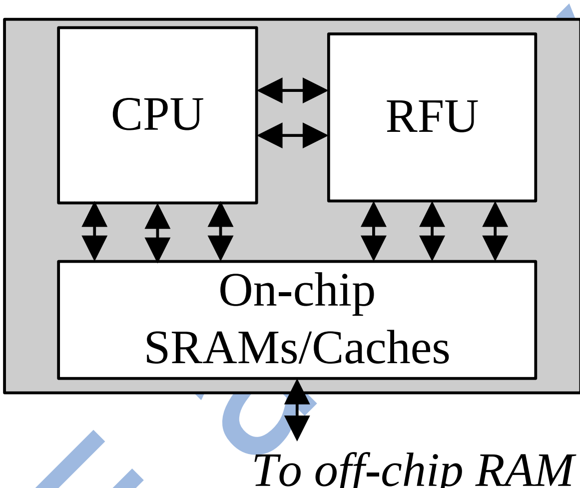

Fig.1. Reconfigurable SoC Architecture.

The RFU is a Coarse-grained Reconfigurable Array (CRA). On-chip memories (SRAMs, caches or combination of them) store program code, CRA configurations and data. Local data and instruction (configuration) memories are located in both the CPU and in the CRA. The CRA acts as a coprocessor to the CPU and accelerates computational intensive software parts of the application. The embedded CPU, typically a RISC like an ARM or MIPS, executes control-dominant sequential parts.

The programming (execution) model of the reconfigurable platform considers that the data communication between the CRA and the CPU uses shared-memory mechanism. The shared memory is comprised of the system's on-chip data RAM and coprocessor data registers inside the RFU. The communication process used by the CPU and the CRA preserves data coherency by requiring their execution to be mutually exclusive. The mutual exclusive execution simplifies the programming since complicated analysis and synchronization procedures are not required.

If we consider the design of IPSec, as we have already mentioned the nested hash function is the limiting factor of its performance. So, the design and implementation of this hash function must be selected to be mapped on the RFU so as to be speeded-up, whereas the rest components can be executed on the CPU illustrated in Fig.1. Moreover certain blocks of SHA-256 hash function, pictured in Fig.2, like padding unit, control unit, message digest extraction etc. can also be assigned for execution on the CPU and not on the FPGA (CRA).

As long as the other basic component of IPSec is concerned (that is AES), from [7], it is derived that AES designs implementations present higher throughputs but also higher operating frequencies. Thus from all points of view SHA-256 is the limiting factor of the performance of the design and implementation of IPSec/IPv6 in reconfigurable Hardware.

Obviously the blocks assigned for execution on the CRA, thus the FPGA, is those which determine the critical path of the incorporated hash function. The critical path of the illustrated architecture is located between the pipeline stages and they are going to be mapped on the FPGA. However in order to boost performance of IPSec, we focus on reducing the critical path of the design mapped on the FPGA, so as to increase performance of the whole system. The optimization of the critical path is solely focused on the operation block, in order to reduce the delay and thus increase the operating frequency.

IV. Proposed Methodology

The generic architecture of a hash function is shown in Fig. 2. Due to the blocks' logic variation from round to round numerous implementations [10, 11, 12], are based on four pipeline stages of single operation blocks. Also from a heuristic survey [11] to hash functions it is clear enough that the best compromise is to apply four pipeline stages so as to quadruple throughput and keep the hash core small as well. This selection was made in the presented methodology as it is shown in Fig.2.

Exploring the generic architecture of Fig. 2 it is easily extracted that the critical path is located between the pipeline stages. The other units, MS RAM and the array of constants, do not contribute due to their nature (memory and hardwired logic respectively), while control unit is a block containing very small counters which also don't contribute to the overall maximum delay. Thus, optimization of the critical path should be solely focused on the operation block.

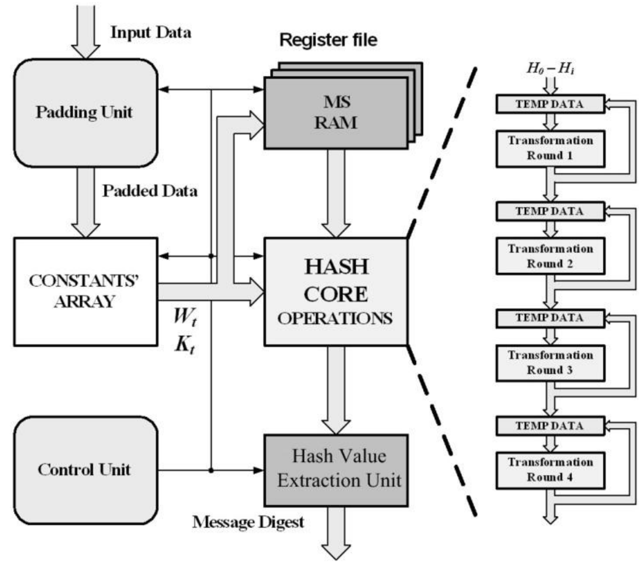

Fig.2. SHA-256 hash core architecture with 4 pipeline stages.

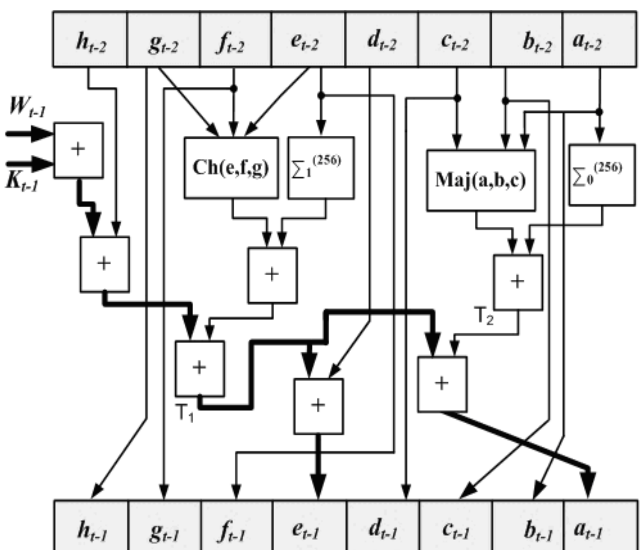

Fig.3. SHA-256 operational block

The operation block of SHA-256 is shown in Fig.3. The critical path (darker line) is located on the computation of at and et values that requires four addition stages and a multiplexer for feeding back the output data.

At the first step of our methodology, a number of operations are partially unrolled. That number is determined by a separate analysis on SHA-256 hash function. This analysis compares variations of partially unrolled operations, their corresponding throughput, the required area and then calculating the proper ratio (cost function). In Fig. 4, the results of a cost function analysis for SHA-256 algorithm, performed in Virtex-II FPGA family, are illustrated. As it is shown, selecting to partially unroll two operations results in the best achieved Throughput/Area ratio (ratio > 2).

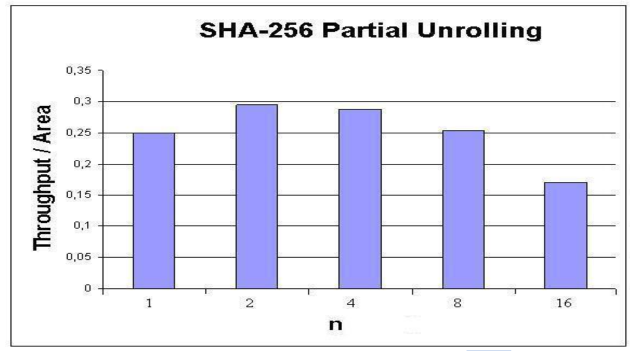

Fig.4. Effect of unrolling the operation blocks of SHA-256

In Fig. 5, the consecutive SHA-256 operation blocks of Fig. 3, have been modified so as to exploit parallel calculations. The gray marked areas on Fig. 5 indicate the parts of the proposed SHA-256 operation block that operate in parallel.

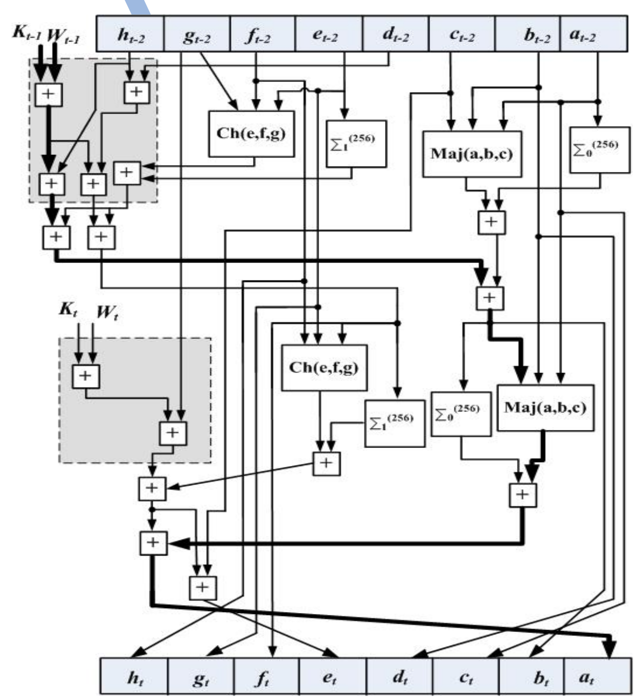

Fig.5. Two unrolled SHA-256 operation blocks.

It is noticed that two single addition levels have been introduced to the critical path that now consists of six addition stages needed for the computation of at and et values. Although, this reduces the maximum operation frequency, the throughput is increased significantly since the message digest is now computed in only 32 clock cycles (instead of 64). The area requirements are increased since more adders have been used in order to achieve the partial unrolling.

The next step of the proposed methodology has to do with the spatial pre-computation technique. Taking into consideration the fact that some outputs are derived directly from some inputs values respectively we can assume that it is possible during one operation to pre-calculate some intermediate values that will be used in the next operation. These pre-calculations are related only with those output values that derive directly from the latter mentioned input values. This pre-computation technique is applied on the partially unrolled operation block in Fig. 5 and the new modified operation block is shown in Fig. 6.

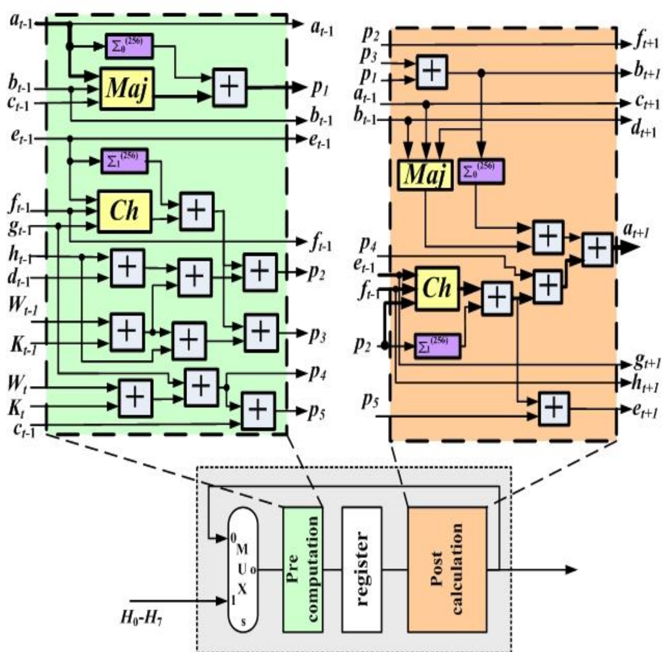

Fig.6. Partially unrolled operation block with pre-computed values.

Observing Fig. 6 it is noticed that the critical path is now located on the computation of the peripheral value p1 that is introduced in Fig. 6. The critical path has been reduced from six addition stages and a multiplexer to four addition stages, two non-linear functions (noted as Maj and Ch in Fig. 6) and a multiplexer. Comparing to the conventional implementation of the single operation block shown in Fig. 3, theoretically throughput has been in-creased by 80%-90%.

This has been achieved by pre-calculating some intermediate values and moving the pipeline registers to an appropriate intermediate point to store them. The new operation block now consists of two units, the "Pre-Computation" unit which is responsible for the pre-computation of the values that are needed in the next operation and the "Post-Computation" unit which is responsible for the final computations of each operation.

The third step of the proposed technique is to apply the system-level pre-computation so as to achieve data pre-fetching. It was noticed that all Wt values can be computed and be available for adequate time before they are really needed in each operation t since they are computed through some XOR bitwise operations. Also the values of the constants Kt are known a priori. These two facts give us the potential of pre-computing the sum Wt + Kt outside of the operation block. The sum is then saved into a register that feeds the operation block and thus the externally (regarding the operational block) pre-computed sum Wt + Kt is available at the beginning of each operation. So at the operational block, from now on it will be assumed that this sum available at the beginning of each operation and its computational time is excluded from the critical path. The new operational block is illustrated in Fig. 7.

Inspecting Fig. 7, we observe that the critical path is located on the computation of the peripheral value p1, and consists of four addition stages and two non-linear functions. However we notice that at the beginning of this path there is the value p4 that is pending to be added to a sum that at the same time is being calculated.

So for this case, a CSA can be used in order to add the three values in advance compared to the necessary time in case we used two adders as in Fig.7. The Carry Save Adder is applied on the "Post-Computation" unit as it is depicted in Fig.8 where we have also used a Carry Save Adder in the "Pre-Computation" unit. This way the critical path inside the operation block has been reduced to one Addition stage, two Non-linear functions and two Carry Save Adders that are required in order to compute the value p1.

The final proposed operation block for SHA-256 is illustrated in Fig. 8. It processes two operations in a single clock cycle, and the critical path is shorter than that of the conventional implementation, resulting in an increase of through-put of more than 110% (theoretical). The introduced area penalty is 3 adders, 4 Carry Save Adders, two 32-bit registers and 2 non-linear functions. The introduced area penalty is about 35% for the whole SHA-256 core compared to the conventional pipelined implementation. This corresponds to an area penalty of about 9% for the whole security scheme. This area penalty is worth paying for about 110% increase of throughput.

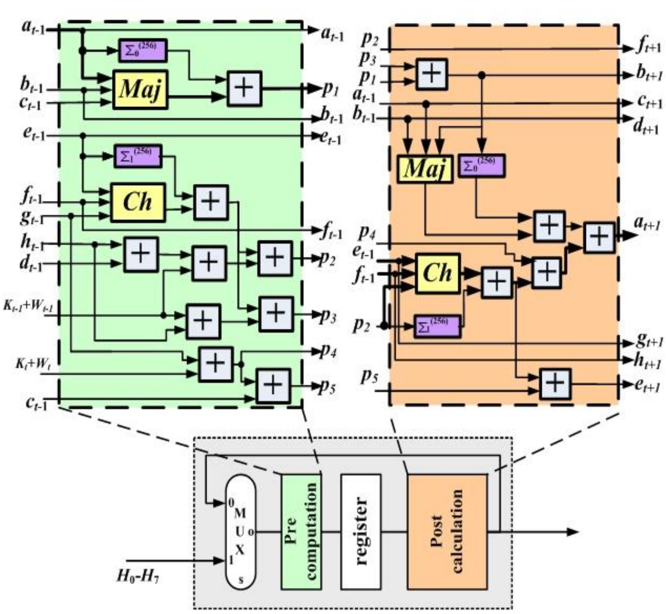

Fig.7. Partially unrolled operation block with pre-computed values for SHA-256 with pre-fetching of W+K values.

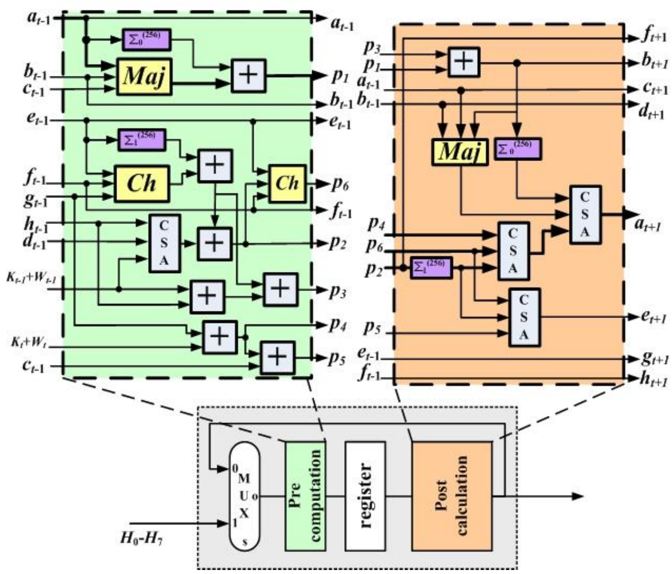

Fig.8. Proposed SHA-256 operation block.

V. Results and Comparisons

In order to evaluate the proposed methodology, SHA-256 hash function was captured in VHDL and was fully simulated and verified.

Table 1: Performance Characteristics and comparisons

SHA-256				
Implementation	Op. Freq. (MHz)	Throughput (Mbps)		
		Post-synthesis	Post Place & Route	Area CLBs
[13] [a]	42.9	77	-	1004
[14] [a]	88.0	87	-	1261
[11] [a]	83	326	-	1060
[15] [a]	82	646	-	653
[16] [a]	77	308	-	1480
[17] [a]	53	848	-	2530
Proposed [a]	**35.1**	**2210**	**2077**	**1534**
[15] [b]	150	1184	-	797
[18] [b]	133	1009	-	1373
[19] [b]	81	1296	-	1938
Proposed [b]	**52.1**	**3334**	**3100**	**1708**
[20 [c]	64	2052	-	1528
Proposed [c]	**36.4**	**2330**	**2190**	**1655**
[21] [d] *(Commercial IP)*	96	-	756	945
[22] [e] *(Commercial IP)*	-	-	1900	1614 (LUTs)
[23] [e] *(Commercial IP)*	133	-	971	asic

[a] *Virtex FPGA family*
[b] *Virtex II FPGA family*
[c] *Virtex-E FPGA family*
[d] *Virtex 4 FPGA family*
e Virtex 5 FPGA family

The XILINX FPGA technologies were selected as the targeted technologies, synthesizing the designs for the Virtex FPGA family.

To exhibit the benefits of applying the proposed design methodology, SHA-256 hash function was implemented following the steps of the proposed methodology and is compared with other existing implementations proposed either by academia or industry.

The results from the latter implementations are shown in Table 1, for a variety of FPGA families. There are reported both post-synthesis and post-place & route results. The reported operating frequencies for the proposed implementations are related to the corresponding post-synthesis results.

As it can be easily seen, the increase observed for SHA-256, is about 110% gain in throughput and 30% area penalty compared to a non-optimized implementation with four pipeline stages (implemented in the same technology).

This way the improvement that arises from the proposed methodology is confirmed and evaluated fairly, verifying the theoretical analysis in the previous section. Furthermore, comparing the implementations of other researchers to those that were resulted from the proposed methodology, it can be observed that all of them fall short in throughput, in a range that varies from 0.75 – 26.4 times less than the proposed implementation.

VI. Conclusions

In this paper a new methodology was proposed for achieving high throughputs for SHA-256 and other hash functions with a small area penalty. The presented methodology is generic and can be used to a wide range of existing hash functions that are currently used or will be deployed in the future and call for high throughputs.

The methodology led to significant increase of throughput (about 110% for SHA-256), compared to corresponding conventional implementations, with a small area penalty. The results derived from their implementation in FPGA technologies confirm the theoretical results of the proposed implementation.

VII. Acknowledgements

This work was supported by action "Young Researchers from Abroad" which is funded by the Cypriot state-Research Promotion Foundation (RPF/IPE).

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Artificial Immune System Implementation upon Embryonic Machine for Hardware Fault-tolerant Industrial Control Applications

Géza HUSI[1] Csaba SZÁSZ[2] Virgil CHINDRIŞ[3]

***Abstract*—Living organisms demonstrate through millions of years evolution remarkably fault-tolerance, robustness and self-healing abilities. Taking inspiration from biological immune systems and embryonic processes which acquire some of these fault-tolerant properties, the paper presents the implementation of an embryonic machine with FPGA-based multicellular architecture, which is able to imitate cells or artificial organism operation mode, with similar robustness and fault-tolerance properties like their biological equivalents from nature. This VLSI hardware structure was upgraded through specially developed algorithms, provided with strongly network communication capabilities and self-healing behaviors. Several casual faults were considered also through the test operations. Detection and localization of these hardware faults were achieved through special reconfiguration operations of active and spare artificial cells inside the embryonic array. Laboratory experiments prove high validity of the considered background theoretical approaches, the developed hardware immune system express remarkable surviving and self-healing capabilities.**

Keyword — artificial cell, embryonic machine, hardware immune system, POE architecture, FPGA circuit

I. INTRODUCTION

As the new generation programmable hardware systems becomes more complex, it becomes increasingly difficult to avoid manufacturing errors or occasional internal faults, and to determine the validity of the system. No electronic components - like diodes or transistors - will function for ever, and these faults can manifest themselves as internal errors, or can ultimately cause a system to fail. Therefore, the ability of a system to function in the presence of faults, and more, to become fault tolerant, is a highly increasing research area for engineers from informatics and microelectronic sciences.

In case of VLSI hardware systems, the traditional fault detection methodologies seem to be very inefficient and also expensive. The alternative for all these problems was inspired from living biological organisms, which are provided with remarkable surviving and fault-tolerance properties. Enhanced with remarkable abilities during a long evolution process, they are under continuous attack from other living entities, can survive infectious pathogens, injury, or several diseases. By adapting these mechanisms and capabilities from nature, scientific approaches have helped researchers understand related phenomena and associated with principles to engine complex novel digital systems and improve their capability. As a result of these efforts, bio-inspired techniques are now frequently used in VLSI digital systems design and development.

In last decade several projects were started on the POE model theoretical background, which means the creation of a hardware system that exhibits learning, evolutionary diversity, and multi-cellular organization [1], [2], [3], [4]. Phylogeny (P) is basically concerned in species genetically evolutionary. In engineering sciences, this corresponds to the genetic algorithms and evolvable hardware. Ontogeny (O) involves multi-cellular organization, cellular division and differentiation from the mother to the daughter cell (each cell owns a copy of the original genome). Finally epigenesis (E) is concerned with learning and adaptation processes (for example: nervous system, immune system) [5], [6], [7].

The embryonic systems were born as a result of the above mentioned research effort, on support of the cellular embryology basic terminology. Usually an embryonic system is considered to be a homogenous array of logic units (called artificial cells), on which is built a hardware multilayered artificial immune system, and which can then accommodate the faults [8]. Such systems, similar to their biological equivalents will posses cellular architecture properties: multi-cellular organization, cellular division, cellular differentiation, and they will virtually mimic every aspect of a living organism to achieve the above mentioned POE features [9], [10].

II. BIOLOGICAL ORGANISMS AS MODEL FOR ARTIFICIAL IMMUNE SYSTEMS HARDWARE IMPLEMENTATION

As it is known, the immune system found in higher evolutional level biological organisms is a distributed and multilayered system that is robust and able to identify infectious pathogens, injury, diseases, or other harmful effects. Therefore, their properties and abilities - like self-healing or surviving - would be more advantageous in many applications were often are imposed robustness and also high security operation requirements. The basic goal of these research efforts is to take inspiration from biological organism's immune system and embryonic processes to acquire these fault tolerant properties in hardware circuits. For this reason, the artificial immune systems have been applied to many different application areas, such as:

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hardware fault tolerance, industrial process monitoring, fault tolerant software, pattern recognition, electrical drives control, neural networks implementation, optimization and industrial control processes [9], [10], [11].

The artificial immune system presented in this paper is modeled on a POE-type embryonic structure developed in analogy with the evolutionary processes of biological systems. In accordance with this model, these embryonic systems derive from the multi-cellular structure of complex living organisms with strong hierarchical organization from molecular to population levels, as shown in figure 1.

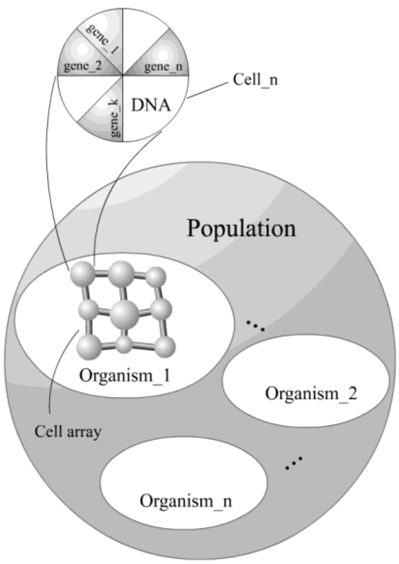

Fig. 1. Living organism's evolution process as model for POE-type artificial immune hardware systems development

All multi-cellular organisms start their life as a single cell, which divides then repeatedly to generate numerous identical copies of itself. Each cell contains all the information necessary to create the entire entity – the genotype, or named DNA, as it is expressed in figure 2.

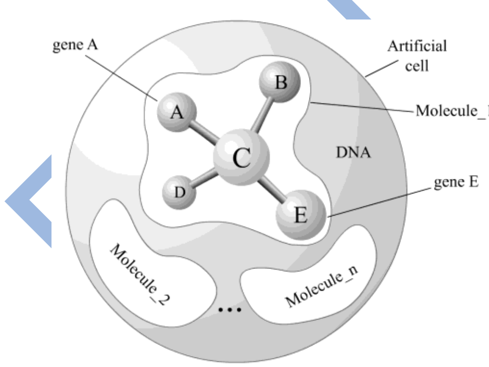

Fig. 2. A single cell structure from a multi-cellular organism

As the number of cells grows, cell differentiation takes place, when some of them start to change to provide different or specialized functionality. In this case the appropriate gene (or genes) is selected based upon the cell's position inside the cell network as well as other factors.

III. FPGA CIRCUITS-BASED ARTIFICIAL IMMUNE SYSTEM MODEL DEVELOPMENT

Considering as starting point and background the theoretical approaches from the previous paragraph, a new model of hardware multilayered artificial immune system based on embryonic array structure is proposed to develop. In this idea, is considered a homogeneous array of programmable logic units (called artificial cells) that use their location within the network to extract appropriate configuration data. Each artificial cell contains all the configuration details of all cells and hence can perform any cell's function as required. Due to avoid complex structures presentation, let's consider a model of entities composed each from 9 cells, organized in macro-groups of cell networks, and named shortly clusters. Figure 3 shows the structure of this conception, limiting the cell network area at first to case of just one cluster.

Fig.3. Artificial cells organized in a cluster structure

For more simplicity, in the model, the cell DNA is designed only with 5 genes (*A*, *B*, *C*, *D*, and *E*), showing active at the same time just one of them (highlighted in the figure). The 4 cells without active genes (dark in the figure) are considered spare cells in the network. The implemented genes are generically labeled in the model with *A*, *B*, *C*, *D*, and *E*, but they can represent in fact a wide range of control algorithms and programs (industrial process control, electrical motors control, etc.) defined by the software implementation.

Previous estimations for hardware implementation seem to evince an increased processing power and network communication abilities for the model. In this context, the FPGAs (Field Programmable Gate Arrays) because of their specific internal architecture circuits seem to be the most appropriate elements for such types of implementation. Thus, it means that each cell will be considered an autonomous FPGA array, with special functions inside the organism, defined through an instruction set (program), and called the cell's gene. Each cell has a copy of all genes from the organism (operative genome), and depending on the cell's position inside the organism, only one cell has an operative gene (the cell's differentiation properties).

By extending the foregoing ideas, it was easy to conclude that the next stage of development should be expanding the previous model with several cell clusters. Thus, it was developed the artificial organism model shown in figure 4.

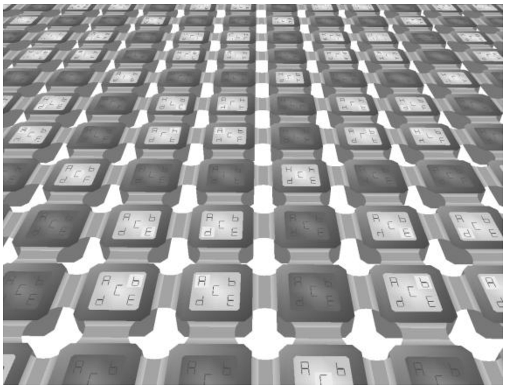

Fig. 4. Artificial organism model

In this structure, there are more interconnected groups (each of 9 cells), which can make up a square matrix architecture or lop-sided with a different number of rows and columns. This setting does not affect in any way the theoretical point of view or model functionality, all artificial cells are going to work together inside a homogenous embryonic array.

Figure 5 gives an example of network structure for the artificial immune system (the cells are numbered after their row and column position).

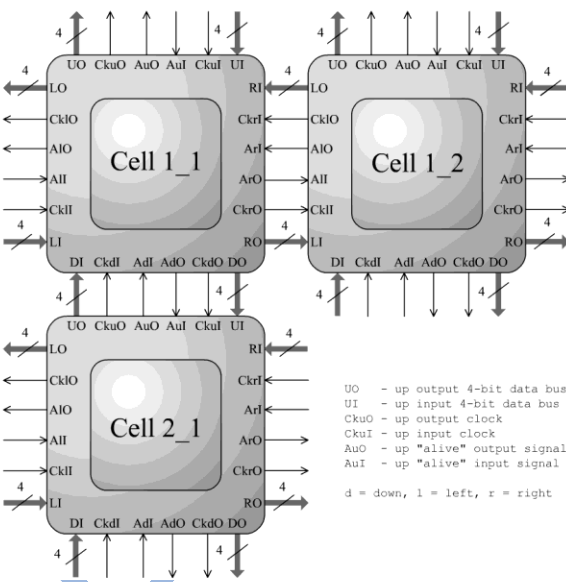

Fig. 5. Network structure of the embryonic system

There are 4-bit data buses on each lattice for source-cluster identification *X* and *Y* coordinates code, destination-cluster identification *X* and *Y* coordinates code, destination- and source-cell identification code, and for the implemented genes code (*A*, *B*, *C*, *D*, and *E*). Cell operation state (active, or faulty) is indicated through *Alive*-type signals (*Alive=1* active, *Alive=0* means faulty cell, e.g.: *AuI=1*), and all communication data are synchronized by *Clk*-type signals.

In figure 6 it is shown a typical fault detection process, where the neighbor cells that discover damage (*Alive=0* on each lattice of one cell) pass through a sequential process this information to entire network. In this way, all artificial cells become aware almost instantaneously of this error or damage.

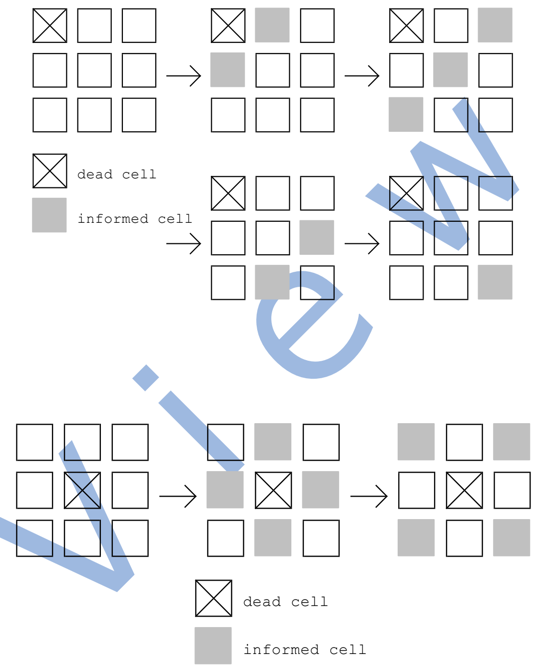

Fig. 6. Detection and information about occurred fault

As it is known, a biological immune system never tries to provide a fault free functionality, typically killing of infected cells. The biological entity can easily accommodate this due to the huge quantity of redundancy inside the organism. In contrast with the above, the hardware immune systems require some inherent fault tolerance, where as many faults inside the system, there are spare cells which can handle the occurred physical errors and damages. These theoretical observations are imitated briefly below in figure 7.

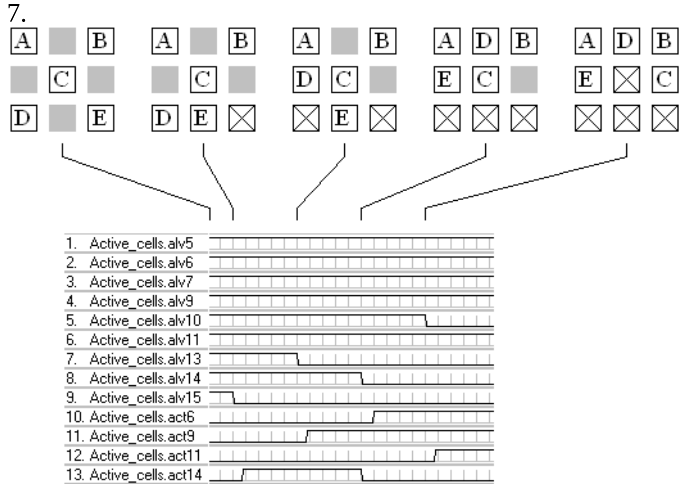

Fig. 7. Immunity of embryonic hardware system

The first cluster from left-upper side of figure, contains 4 spare cells (the cells without active gene) that are not utilized until a fault occurs. In case of fault, for example the cell which shows gene *E*, all buses on the 4 lattice of the cell are released and thus the neighboring spare cell takes over the functionality of faulty cell. It should be reiterated that no configuration data has to be recalculated or moved; just the change in coordinate is all that is required for the cell to reconfigure themselves. This elimination process can be repeated more and more, until are not enough spare cells inside the cluster. But in any case it can be observed that the embryonic array keeps during this process its immunity, showing active the same genotype (A, B, C, D, and E). The faulty cells replacing process with spare cells is presented also through the time diagram of Alive signals shown below the considered clusters. The above presented methodology can increase considerably the fault tolerance and self-healing properties of the embryonic array, and is well suitable for reconfigurable hardware implementation.

IV. Artificial Immune System Simulation

As it is known, an immune system is composed of a huge number of cells which protects an organism from infection and pathogens. From this point of view biological organisms are observed to have an amazing stability. The cost to have such properties is huge; a reliable embryonic system usually has redundant information about itself. These observations are briefly illustrated also in figure 8, where an artificial organism (composed by 6 cell clusters) with multiple faults is considered for computer-aided simulation.

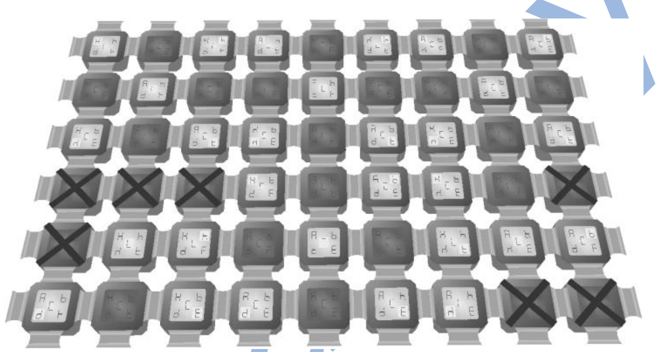

Fig. 8. Multiple faults in the artificial organism

Several alleatory faults of active or spare cells are considered in this simulation example. Faulty of redundant cells (spare, or without active gene) do not mean in any manner a threat for the organism functionality. But the faulty of one active cell, for example with gene B in cluster 2_1 (left-down side of the organism) is a real problem. As it is observed from the figure, this out of order cell was replaced already by a redundant spare cell which shows the same active gene B (highligted in the picture). In the cluster 2_3 (right-down side) three cells are faulted one after the other: one spare cell, and two active with genes B and E. After these unwanted events, from the four spare cells of the cluster, two cells become instantaneuosly active showing the same gene B and E. The result is: the cluster 2_3 kept its immunity, showing active the same genotype (A, B, C, D, and E), and remains immune, with high fault-tolerance ability. If the clusters can maintain individually their high immunity (in strong relationship with the available number redundant or spare cells), it means that the whole organism structure is also protected against any occasional faults. These events are presented also by a signals time-diagram (the *Alive* signals of each cells in the mentioned clusters), as given in figure 9. A huge number of similar computer-aided simulations is possible to investigate using the artificial organism model presented already in figure 4. In each case, the result was the same robustness and fault-tolerance, proving the viability of the developed artificial immune system model presented in the previous paragraph.

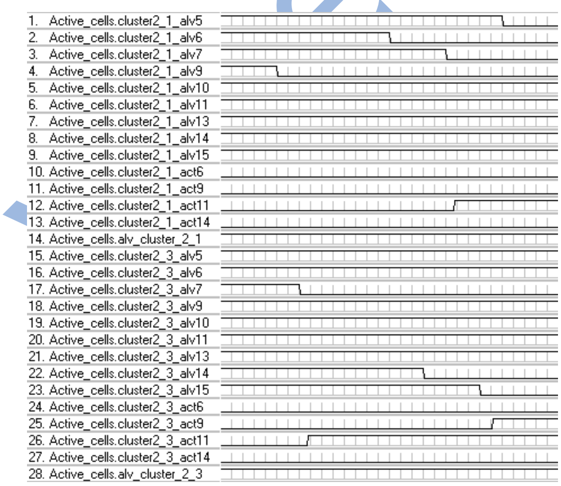

Fig. 9. Time-diagram of multiple faults in the cluster

It is important to mention, that there are important differences between natural biological systems and human-made systems. In the human-made systems, if a small piece does not operate properly, the system is not very reliable. In biological systems even an essential fault cannot stop operability of the full system because information about structure of the whole system and algorithms to self-healing are kept in a decentralized manner. The artificial hardware immune system model presented here follows also the above mentioned decentralization strategy.

V. Artificial Immune System Implementation On FPGA-Based Reconfigurable Hardware

The hardware implementation part of the project is motivated by two main goals: to perform a preliminary study of the ability of the presented artificial immune system model to imitate as close as possible the living organism's remarkable adaptation, surviving and fault-tolerance properties, and to develop fast implementation to explore the reconfigurable architecture hardware systems flexibilities. To use FPGAs is perfect for such an application, because the computation involved in the system is a massively parallel problem. Furthermore, an FPGA-based system can be easily reconfigured to test a several theoretical approaches and hypotheses [12].

For implementation purpose it is defined at first the FPGA circuit-based artificial cell hardware structure (figure 10). This operation starts from consideration that each artificial cell from the presented models is built on using an autonomous FPGA array. Basic communication rules between cells inside the network are defined after the model described in paragraph III, and presented in figure 5.

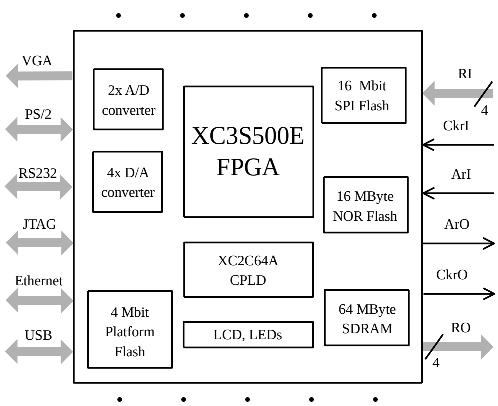

Fig. 10. Artificial cell block diagram

The proposed implementation, which is synthesized in an FPGA XC3S500E circuit, employs reconfigurable pipeline, 4 Mbit Platform Flash Memory, 16 Mbit SPI Flash, 16 MByte NOR Flash, 64 MByte DDR SDRAM Memory, 64-macrocell XC2C64A CoolRunner CPLD, SHA-1 1-wire serial EEPROM, 4-output SPI-based D/A Converter (DAC), 2-input SPI-based A/Digital Converter (ADC), and ChipScope™ SoftTouch debugging port [13]. To communicate with external devices is available for users also one PS/2 port, VGA display port, 10/100 Ethernet PHY, two RS-232 ports, and USB-based FPGA/CPLD download/debug interface. A photography overview of the Spartan-3E Starter Kit development board built on the above presented FPGA XC3S500E circuit is shown in figure 11 [12], [13].

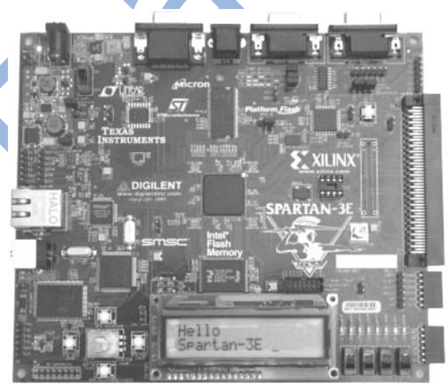

Fig. 11. Spartan-3E Starter KIT development board [13]

It's important to mention that the hardware system development strategy follows some special purpose requirements and implementation guidance. One of them is to avoid artificial cells hardware resources and memory utilization for initialization, configuration, or other auxiliary routines and algorithms implementation. The main goal is to maintain all their capability for complex software processing by algorithms that are able to imitate with high fidelity the biological organism's sophisticated adaptation, robustness, and immunity properties. This is the basic reason of choosing one stand-alone FPGAs for each artificial cell, with the above mentioned powerful hardware and software resources built in an FPGA XC3S500E circuit. The block diagram given in figure 12 shows this implementation strategy.

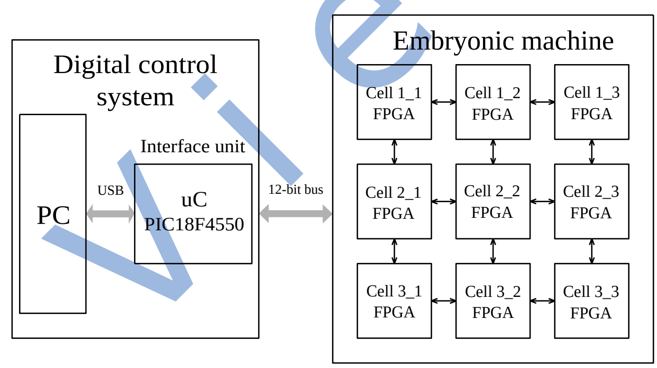

Fig. 12.The hardware system block diagram

The main goal was to design and construct a versatile framework system. There are two basic novel hardware structures: in the right side, it is depicted the developed embryonic machine (hardware immune system) with its array structure, and in the left the supervisor digital control system. No other functions or tasks are executed by the embryonic machine, just the regarded network communication abilities and specially developed fault-tolerance algorithms, in order to reproduce artificial hardware immune system behaviors. All auxiliary functions or drivers like interfacing, initialization, or data acquisition are processed by the supervisor digital control system, built on a personal computer and PIC18F4550-type microcontroller. A general view of the laboratory experimented test system is shown in figure 13.

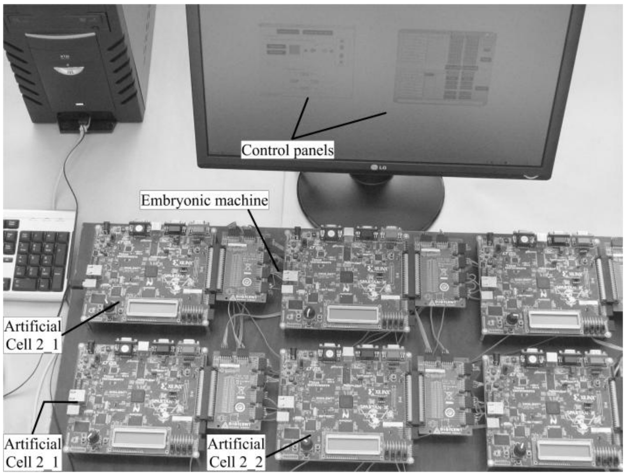

Fig. 13. The embryonic system

Data acquisition operations and initialization tasks are performed by the PIC18F4550 controller which operates like an intelligent interface unit between the personal computer and embryonic array. The main role of the computer consists of monitoring the network communication and data transfer rate inside the artificial cell network, through specially designed software control panels. Also each occurred fault in the embryonic machine is detected in real-time and represented in the panel's window.
The experimental results are focused upon monitoring the communication waveforms and data transfer protocol between the personal computer and embryonic machine via PIC18F4550 microcontroller. In this way there are checked continuously the right interoperation tasks between artificial cells, information about faulty cells and their replacing process, the cells coordinate detection inside the network, and the adequate genotype transmission for neighboring cells. The first example from figure 14 presents through the most relevant signal waveforms the above mentioned network operations time-diagram.

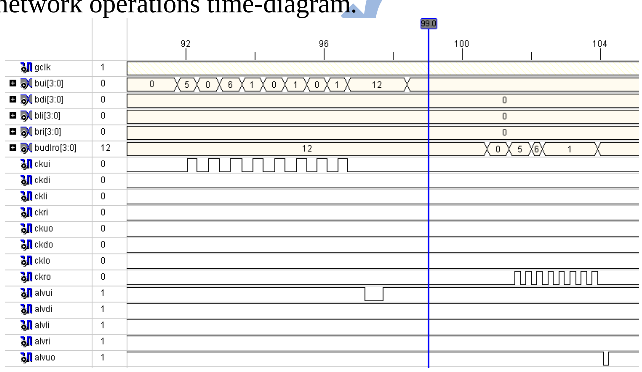

Fig. 14. Network communication waveforms (example_1)

In left-side of figure there are shown the 8 clocks of the *ckui* (*clk up in*, means PIC18F4550 controller connected to the upper side of the cell) signal which synchronizes the data transfer through the *bui(3:0)* data bus from controller to cell 1_1. This cell processes the captured instructions, and via the *alvui* signal marks the end of this operation. In right-side of figure cell 1_1 passes toward the captured instructions to cell 1_2, which in this example performs the position of destination cell.

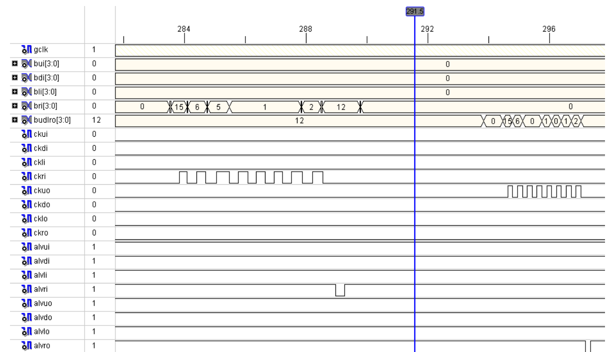

Fig. 15. Network communication waveforms (example_2)

In the second example from figure 15, cell 1_2 is already in role of the source cell, uploaded before with the data packages mentioned in example_1. After data processing over, cell 1_2 returns the result of computing via cell 1_1 to the interface unit built with the PIC18F4550 controller. In this case, the signals involved in communication are *ckri*, which synchronizes the data (signal *bri*) received by cell 1_1 from cell 1_2, and *ckuo* which synchronizes the data (signal *budlro*) from cell 1_1 to personal computer. The data transfer rate on the laboratory experimented embryonic machine was programmed during tests sufficiently low (3-5Hz) to ensure all network communication operations visual observation.

VI. Conclusions

The developed artificial hardware immune system may become helpful support for future developments in embryonic systems, in order to founding the theoretical basis, design models or development methods of this relatively new science domain named embryonic systems. The presented theoretical approaches and proposed models were carefully tested and implemented on a new generation of FPGA-based development system with a generous hardware resource with high-level programming and upgrading possibilities.
The importance of these research efforts consist basically of the influence of embryonic systems regarding the evolution of other engineering sciences from microelectronics and informatics. One of the most spectacularly application of embryonic systems are in high performance industrial control processes, where the system hardware security and maintenance is a major criteria. Not at last, through implementing the basic properties of living organisms on digital systems it becomes possible to realize high performance fault-tolerant and self-healing hardware architectures.

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Fast Fractal Image Compression Based on Domain-Range Pixel Value Difference

Venkata Rama Prasad Vaddella[1] Ramesh Babui[2]

Abstract- Fractal image Compression **is a lossy compression technique that has been developed in the early 1990s. It makes use of the local self similarity property existing in an image and finds a contractive mapping affine transformation (fractal transform) T, such that the fixed point of T is close to the given image in a suitable metric. It has generated much interest due to its promise of high compression ratios with good decompression quality. The other advantage is its multiresolution property, i.e. an image can be decoded at higher or lower resolutions than the original without much degradation in quality. However, the encoding time is computationally intensive. In this paper, a new method to reduce the encoding time based on computing the pixel value difference of domain and range blocks is presented. A comparison for best match is performed between the domain and range blocks only if the range block pixel value difference is less than the domain block pixel value difference. This reduces the number of comparisons, and thereby the encoding time considerably, while obtaining good fidelity and compression ratio for the decoded image. Experimental results on standard gray scale images (512x512, 8 bit) proved that the proposed method improved in performance when compared to conventional fractal encoding.**

*Keywords:*Fractal image compression, pixel value difference, adaptive scaling, classification.

I. Introduction

The basic scheme of fractal image compression is to partition a given image into non overlapping blocks of size rxr, called range blocks and form a domain pool containing all of possible overlapped blocks of size 2rx2r, called domain blocks associated with 8 isometries from reflections and rotations [19]. For each range block, it exhaustively searches, the domain pool, for a best-matched domain block with the minimum rms error after applying a contractive affine transform to the domain block. A fractal-compressed code for a range block consists of quantized contractivity coefficients in the affine transform, a luminance offset, the position of the best-matched domain block and its isometry. The decoding is to find the fixed point, the decoded image, by starting with any initial image. The procedure applies a compressed local affine transform on the domain block corresponding to the position of a range block until all of the

decoded range blocks are obtained. The procedure is repeated iteratively until it converges. The problems that occur in fractal encoding are the computational demands and the existence of best range-domain matches [20]. The most attractive property is the resolution-independent decoding property. The image can be decoded at an enlarged size so that the compression ratio may increase exponentially [18]. However searching the domain pool is highly computationally intensive. For an nxn image, the number of range blocks are (nxn/rxr) and the number of domain blocks are (n-2r+1) x (n-2r+1). The computation of best match between a range block and a domain block is O (r2). If r is constant, the computation complexity of entire search is O (n4).

Yuval Fisher [18] proposed the quad tree-partitioning algorithm for fractal image compression. In this algorithm, the range blocks and domain blocks are classified in to 3 major classes based on the average of the pixels in four quadrants of the blocks. These are further divided in to 24 sub classes (! 4) based on the variance of the pixels in the four quadrants. Thus, the domains and ranges are classified in to a total of 72 classes. This algorithm is called classified search algorithm, as the domains and ranges belonging to the same class only are compared. But due to the large number of domains, the encoding time is very high. One of the simplest ways of decreasing coding time is to reduce the size of the domain pool. This is achieved by a spatial constraint on the domain pool for each range to which it is mapped [20]. Noting that a contractive mapping requires a domain with a higher variance than the range, domains with low variance may be excluded from the domain pool [5]. Alternatively, domain pools may be pruned in order to eliminate domains, which have similar invariant representations to other domains in the pool [15]. During the last decade several researchers have proposed methods to reduce the size of the domain pool based on various split decision functions [11]. The variance feature has been used [4,5,12] as a decision function by many researchers for domain pool reduction. Recently, the entropy function has also been reported as a split decision function [2] to reduce the domain pool. Tomas Zumbakis and Jonas Valantinas [23] have proposed an approach to improve the encoding times based on the classification of the range and domains based on their smoothness estimates in the frequency domain. Daniel Riccio and Michele Nappi [1] proposed a method for reduction of the encoding time by deferring the range and domain comparisons with respect to a preset block. In this paper, we present a new method for reducing the encoding times based on computing the pixel value difference of the domain and range blocks. The comparison

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for a best match between a range and domain pair is then made only if the pixel value difference of the range block is less than the domain block pixel value difference. Quadtree partition algorithm is used [18] for partitioning the image. The domain and range classification is done based on the mean and variance.

II. Fractal Image Compression

Initially, the given image of size nxn is partitioned into overlapping domain blocks Di (of size 2rx2r), for each quadtree partition, where rxr is the size of the range blocks Ri. The domain step size used is δh=δv=4 in horizontal and vertical directions. The domains are classified based on the mean and variance of the pixels in the four quadrants of the block [18]. The domain pool D (search codebook) is constructed by placing the entire domain blocks Di, corresponding to same class in individual lists. The range-domain matching process consists of contracting each domain block to the size of the range block by averaging 2x2 pixels. During encoding, a potential range Ri, is also classified. The domain range matching process consists of searching the domain pool D for the Di and an affine transformation wi, which minimizes the rms distance between the range block Ri and the transformed domain block wi.Di, (i.e. wi .Di ≈ Ri). For a range block with n pixels, each with intensity ri and a decimated domain block with n pixels, each with intensity di, the objective is to minimize the quantity,

$$E\,(R_i, D_i) = \sum_{i=1}^{n} (s.d_i + o - r_i)^2 \qquad (1)$$

occurs when the partial derivatives with respect to s and o are zero. Solving the resulting equations will give the best coefficients s and o [5].

$$s = n\frac{\left[\sum_{i=1}^{n} d_i r_i - \sum_{i=1}^{n} d_i \sum_{i=1}^{n} r_i\right]}{\left[n\sum_{i=1}^{n} d_i^2 - \left(\sum_{i=1}^{n} d_i\right)^2\right]} \qquad (2)$$

$$\frac{1}{n}\left[\sum_{i=1}^{n} r_i^2 + s\left(s\sum_{i=1}^{n} d_i^2 - 2\sum_{i=1}^{n} d_i r_i + 2o\sum_{i=1}^{n} d_i\right) + o\left(n.o - 2\sum_{i=1}^{n} r_i\right)\right]$$

$$n\sum_{i=1}^{n} d_i^2 - \left(\sum_{i=1}^{n} d_i\right)^2 = 0,$$

then s= 0, and $o = \frac{1}{n}\sum_{i=1}^{n} r_i$

the rms error, erms $= \sqrt{E(R_i, D_i)}$ (3)

III. Proposed Method

An improved fractal image compression scheme based on the difference of pixels with maximum and minimum intensity values in the domain and range blocks is proposed. During the encoding phase, the range blocks with pixel value difference less than the pixel value difference of the domains are compared for further regression analysis (for a match). An adaptive parameter β is defined (range between 1.0 to 2.0) for scaling the pixel value difference of a domain block in different quad tree partitions. A significant reduction in encoding time is expected.

Pixel value difference

Consider a single pixel in a domain block Dj. The affine transformation in fractal encoding maps its pixel value pi to the range block Rj, using the equation,

pi (Rj) = s. pi(Dj) + o (4)

The contrast scaling parameter s must satisfy the condition 0<s<1. Let the maximum and minimum intensity level values of the pixels in a generic square block, B, are respectively, pmax (B) and pmin (B). The pixel value difference of the block B, is defined by the relation,

pdiff (B) = pmax (B) - pmin (B) (5)

Using the equations (4) and (5),

pdiff (R) = pmax (R)- pmin (R) (6)

= {s. pmax(D) + o}- {s. pmin (D)+ o}

= s.{ pmax (D) – pmin (D) }

= s. pdiff(D) (7)

Considering the contrast scaling requirement, 0<s<1, equation (7) can be written as,

pdiff (R) < pdiff (D) (8)

In the proposed implementation, for achieving better results, equation (8) is written as,

pdiff (R) < β. pdiff (D) (9)

Where, β, is an adaptive scaling parameter (varying between 1.0 and 2.0) for each quad tree partition. The condition given in equation (9) provides an effective decision rule to avoid an improper domain and range match. Only, domains satisfying the above condition will be compared for the regression analysis. Thus, many unqualified domains are avoided from comparison.

Adaptive scale parameter β for domain block pixel value difference

The parameter β is chosen adaptive for each quadtree depth, *i* to scale the pixel value difference of the domain blocks. For quadtree depth 0, (corresponding to min_part), β_0 is assigned a small initial value (in the present work, β_0=1.25). For other quadtree depths, the scale parameter is computed using the formula, β_{depth}=1.25*$\beta_{depth-1}$. This equation is fit, by conducting repeated experiments on images of different sizes and textures, testing for optimal value of encoding time, quality and compression ratio.

IV. Proposed Algorithm

Step 1: Construct the domain pools $\mathbf{D}_{depth}$, corresponding to each quad tree partition level starting from minimum partitions to maximum partitions (depth=0 to max_part-min_part).

Step 2: Calculate the block pixel value difference using equation (5) of all the domain blocks in each pool $\mathbf{D}_{depth}$.

Step 3: Classify and sort the domains in each pool $\mathbf{D}_{depth}$ in ascending order of the pixel value difference, and place on a list structure.

Step 4: Search for a best match between a range and domain belonging to the same class.

```
write_header_info; (min_part, max_part, domain_step, hsize, vszie)
 depth=0; e_c =rms_tol;
  Function Quadtree(image, depth) {
best_rms=infinity;
β0=initial value; βdepth=1.25* βdepth ;
        While (depth<min_part) Quadtree (image, depth+1);
    Set R1 = I2 and mark it uncovered.
    While there are uncovered ranges Ri do {
//Select the domain pool list Ddepth
 Corresponding to the current range block Ri.
            For (j=1; j<num_domains; ++j) {
              If  (Rpdiff  < β *Dpdiff) {
                Compute s, o, sym_op;
 Compute E(Ri, Di);
  If E (Ri,Di ) ≤ best_rms {
                  best_rms= E(Ri, Di );
    best_domain=(domain_x,domain_y)
 }
              }// End for num_domains
If (best_rms>ec) and (depth<max_part)
                      Quadtree (image, depth+1);
            Else
            Write_transformations  (best_domain,  s,  o, sym_op);
     }// End while uncovered ranges
 }// End function Quadtree()
```

V. Experimental Results

In this section, results of the experiments conducted on various images (512x512, 8 bit gray scale) are presented. The results are compared with Fisher's classified search method [18].

The following values are used for various parameters:

5 bits were used to quantize the scaling coefficient s, and 7 bits for the offset, o.

For all images, the maximum range size is 16x16 (minimum quadtree depth 5), and the minimum range size is 4x4 (maximum quadtree depth 7). Three levels of quad tree partition are used.

The domain pool is constructed with a domain skip distance, δ_h=4 and δ_v=4, i.e. the distance between adjacent domains is 4 pixels.

The rms error tolerance, e_c is given values of 1,4,8,10,15, and 20, leading to results ranging from low to high compression. PSNR is computed after post processing.

Encoding Parameters

The following values are assigned for other parameters (Common to all images).

Image size: 512x512 (8 bit gray scale)

Number of quad tree partitions = 3

Total Number of Domains:

Three different sizes of domains are computed, corresponding to the three quad tree partitions.

Size 32x32 = ((512-32)/4+1)*((512-32)/4+1)= 14,641

Size 16x16 = ((512-16)/4+1)*((512-16)/4+1)= 15,625

Size 8x8 = ((512-8)/4+1)*((512-8)/4+1) = 16,129

Total number of domains in all partitions = 46,395

In the proposed method, the adaptive parameter β (for scaling the domain block pixel value difference) is assigned an initial value, β_0=1.25, and β_{depth} = 1.25* $\beta_{depth-1}$.

The algorithm is implemented in C language, using VC++6.0 compiler. Execution is carried out on a Personal Computer with Intel Centrino Duo T2250 processor with clock frequency @1.73 GHz, with 1.0 GB of RAM.

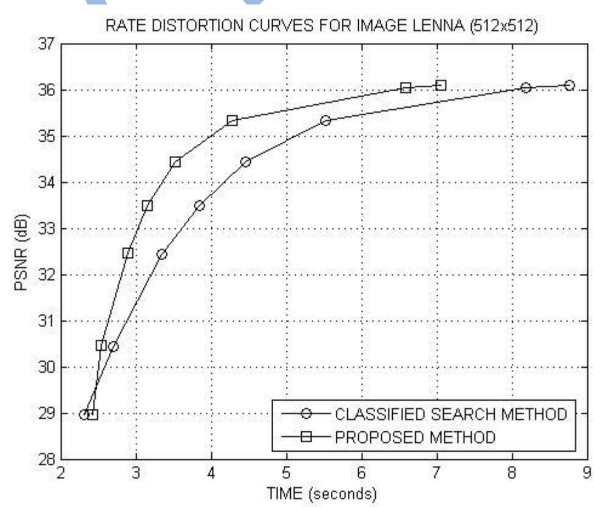

Fig.1. Encoding time vs. PSNR for Lenna Image

TABLE I

RESULTS ON IMAGE LENNA (512X512, 8 BIT)

Classified Search Method			Proposed Method		
CR	Time (sec)	PSNR (dB)	CR	Time (sec)	PSNR (dB)
4.36	8.76	36.09	4.36	7.06	36.09
4.85	8.18	36.04	4.85	6.59	36.04
8.67	5.53	35.34	8.66	4.28	35.34
11.96	4.46	34.43	11.96	3.53	34.43
15.46	3.84	33.49	15.45	3.15	33.49
19.13	3.35	32.44	19.08	2.89	32.46
29.23	2.70	30.43	29.13	2.54	30.46
41.91	2.31	28.95	41.71	2.42	28.96

TABLE II

Results On Image BABOON (512x512, 8 bit)

Classified Search Method			Proposed Method		
CR	Time (sec)	PSNR (dB)	CR	Time (sec)	PSNR (dB)
4.36	7.45	25.49	4.36	5.79	25.49
4.36	4.36	25.49	4.36	5.79	25.49
4.45	7.26	25.49	4.44	5.75	25.49
4.92	6.76	25.45	4.92	5.26	25.44
5.44	5.26	25.37	5.44	4.79	25.36
5.96	5.82	25.22	5.95	4.43	25.22
7.30	5.00	24.64	7.27	3.75	24.66
9.14	4.25	23.65	9.02	3.23	23.74
19.58	2.70	21.28	18.50	2.28	21.44

TABLE III
Results on Image GOLDHILL (512x512, 8 bit)

Classified Search Method			Proposed Method		
CR	Time (sec)	PSNR (dB)	CR	Time (sec)	PSNR (dB)
4.37	9.01	33.87	4.37	7.56	33.85
4.61	8.72	33.86	4.61	7.29	33.84
5.11	8.09	33.72	5.11	6.68	33.71
6.76	6.75	32.90	6.76	5.62	32.89
9.15	5.62	31.75	9.14	4.57	31.75
12.50	4.52	30.64	12.43	3.85	30.66
25.63	3.01	28.31	25.44	2.81	28.33
43.55	2.29	26.87	42.52	2.39	26.92

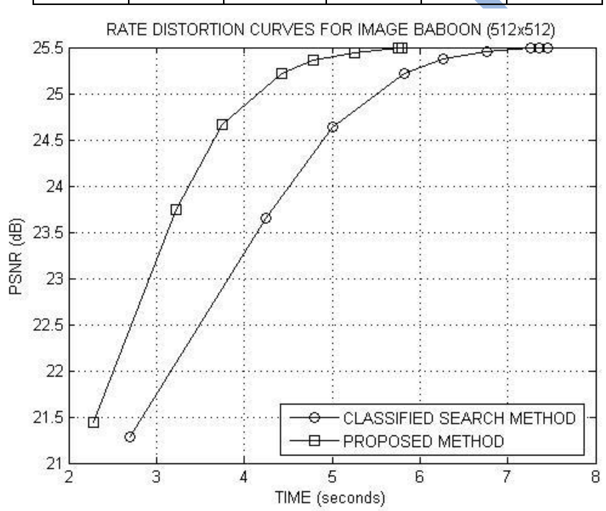

Fig.2. Encoding time vs. PSNR for Baboon Image

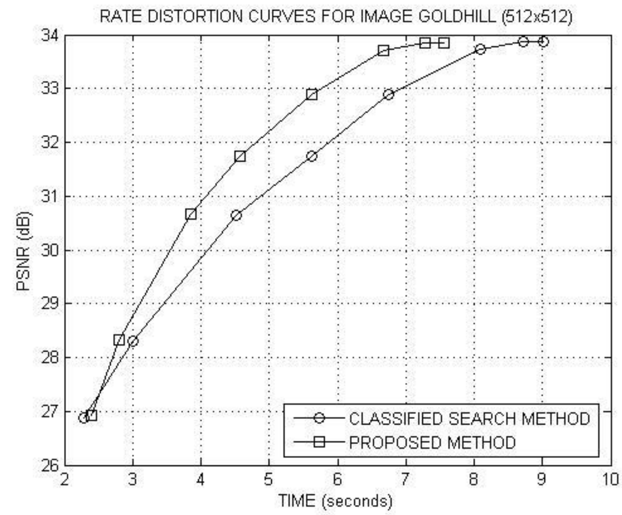

Fig.3. Encoding time vs. PSNR for Goldhill Image

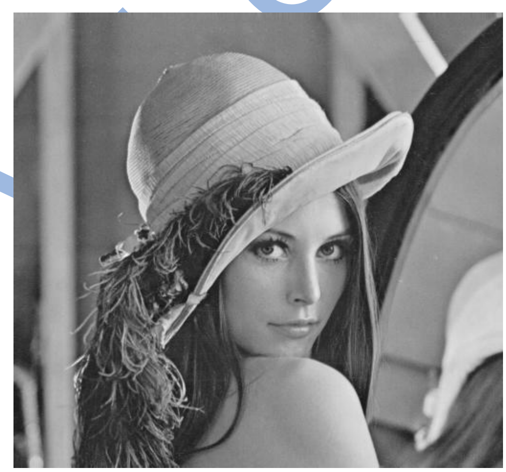

Fig.4.Original Image of Lenna (50%, 512x512, 8bit)

TABLE IV
RESULTS ON IMAGE PEPPERS (512X512, 8 BIT)

Classified Search Method			Proposed Method		
CR	Time (sec)	PSNR (dB)	CR	Time (sec)	PSNR (dB)
4.36	9.64	34.83	4.36	7.48	34.83
4.44	9.48	34.83	4.44	7.46	34.83
6.93	7.18	34.43	6.93	5.42	34.40
11.62	5.23	33.55	11.59	4.09	33.53
16.53	4.25	32.75	16.49	3.43	32.72
21.14	3.70	31.90	21.09	3.12	31.88
32.92	2.84	30.19	32.78	2.75	30.19
45.64	2.40	28.72	45.04	2.50	28.76

Fig.5. Decoded Lenna Image by proposed method(CR=41.71, PSNR=28

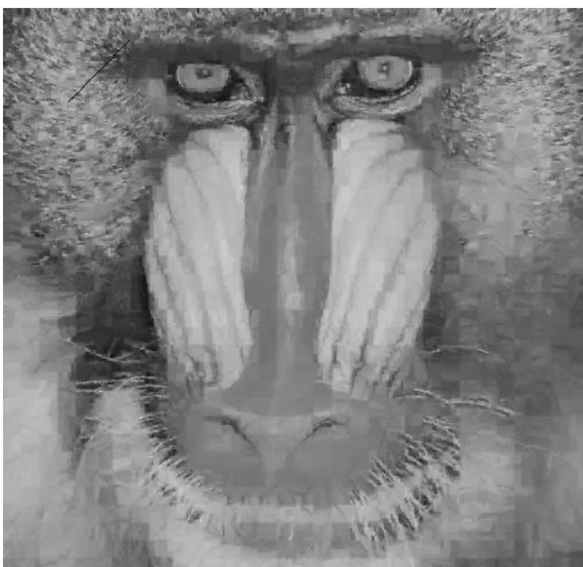

Fig.6. Decoded Baboon Image by proposed method (CR=18.50, PSNR=21.44)

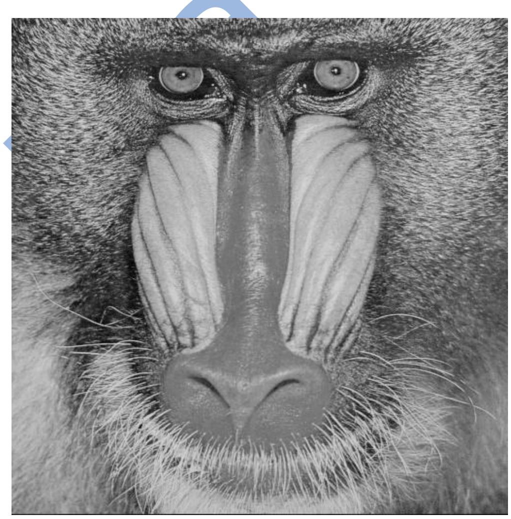

Fig.7.Original Image of Baboon (50%, 512x512, 8bit)

Fig.8.Original Image of Goldhill (50%, 512x512, 8bit)

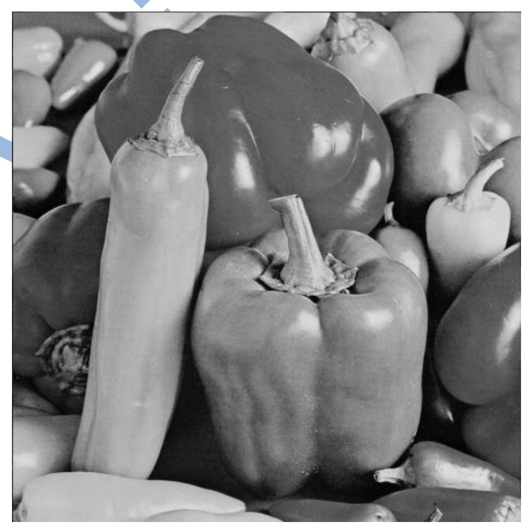

Fig.9.Original Image of Peppers (50%, 512x512, 8bit)

Fig.10. Decoded Image Goldhill by proposed method (CR=42.52, PSNR=26.92)

Fig.11. Decoded Image Peppers by proposed method (CR=45.04, PSNR=28.76)

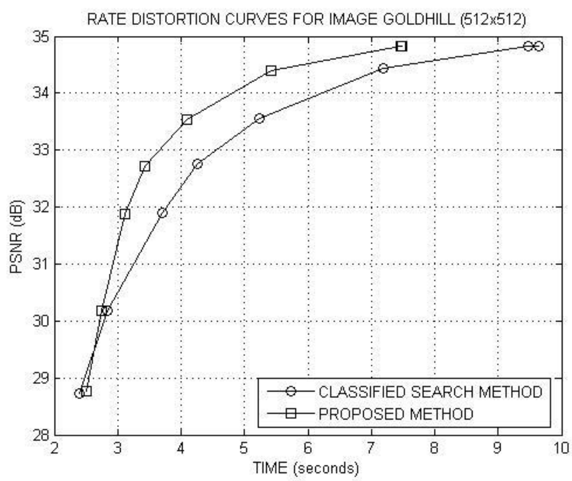

Fig. 12. Encoding time vs. PSNR for Image Peppers

VI. Conclusions

In this paper, an improved classified search algorithm for fractal compression algorithm based on adaptive pixel value difference technique is proposed. Experimental investigations revealed that the method reduces the encoding time significantly when compared to traditional classified search algorithm [18]. The reduction in PSNR is 0.05dB for peppers image. The reduction in compression ratio (CR) is by a factor of 1.73 for gold hill image.

VII. Acknowledgement

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Mitigating Qos Routing Challenges In Mobile Ad Hoc Networks Considering Lifetime And Energy Predictions With Traffic Distribution

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***Abstract-* Mobile Ad hoc Networks are highly dynamic networks. Quality of Service (QoS) routing in such networks is usually limited by the network breakage due to either node mobility or energy depletion of the mobile nodes. To fulfill certain quality parameters, and to achieve network stability, presence of multiple node-disjoint paths becomes essential. Such paths aid in the optimal traffic distribution and reliability in case of path breakages. To maintain such stability requires that links. To cater such problem, we present a node-disjoint multipath protocol. The metric used to select the paths takes into account the stability of the nodes and the corresponding links, calculated through their position and the energy drain rate. Optimal paths are also selected and the load is distributed proportionally to avoid overburden on the nodes .The proposed technique is also illustrated with an example and compared with another similar protocol ENDMR using ns-2.**

Keywords-QoS Routing; Mobile Ad hoc Network; Energy-Aware Routing; Multipath Rrouting, Node-disjoint Routing

I. Introduction

A Mobile Ad Hoc Network (MANETs) [1, 2] is collection of mobile/semi mobile nodes with no existing pre-established infrastructure, forming a temporary network. Such networks are characterized by: Dynamic topologies, existence of bandwidth constrained and variable capacity links, energy constrained operations and are highly prone to security threats. Due to all these features routing is a major issue in ad hoc networks. The traditional routing protocols for ad hoc networks, classified as Proactive/table driven e.g. Destination Sequenced Distance Vector (DSDV) [3], Optimized Link State Routing (OLSR)[4], Reactive/On-demand, e.g. Dynamic Source Routing Protocol (DSR) [5] , Ad hoc On-Demand Distance Vector routing protocol (AODV) [6], Temporally Ordered Routing Algorithm (TORA)[4] and Hybrid, e.g. Zone Routing Protocol (ZRP) [7], Hybrid Ad hoc Routing Protocol (HARP) [23], attempt to provide only best effort delivery. Their target is limited to finding the minimum hops or the shortest paths.

Quality of Service (QoS) based routing is defined in RFC QoS

2386 [8] as a "Routing mechanism under which paths for The main objectives of QoS based routing are[8]:Dynamic determination of feasible paths for accommodating the of the given flow under policy constraints such as path cost, provider selection etc, optimal utilization of resources for improving total network throughput and graceful performance degradation during overload conditions giving better throughput. QoS routing strategies are classified as source routing, distributed routing and hierarchical routing [9]. QoS based routing becomes challenging in MANETs, as nodes should keep an up-to-date information about link status. Also, due to the dynamic nature of MANETs, maintaining the precise link state information is very difficult. Finally, the reserved resource may not be guaranteed because of the mobility-caused path breakage or power depletion of the mobile hosts. QoS routing should rapidly find a feasible new route to recover the service. Our motive in this paper is to design a routing technique, which considers all three above problems together. We define a metric that attempts to maintain a balance between mobility and energy constraints in MANETs. We use Dynamic Source Routing (DSR) [5], as the base protocol to design our model. The designed technique is compare to a similar protocol Energy Aware node Disjoint Routing (ENDMR) [17] using ns-2 simulator.

II. Related Works

In the recent period lot of research has been done in QOS based, multi-path and node disjoint routing. Lately, the upcoming concern is the energy issues in mobile ad hoc networks (MANETs) The recent studies extensively focused on the multipath discovering extension of the on- demand routing protocols in order to alleviate single-path problems like AODV[6] and DSR[5], such as high route discovery latency, frequent route discovery attempts and possible improvement of data transfer throughput. The AODVM (AODV Multipath) AOMDV [10] , is a multipath extension to AODV. These provide link-disjoint and loop free paths in AODV. Cross-layered multipath AODV (CM-AODV) [11], selects multiple routes on demand based on the signal-to-interference plus noise ratio (SINR) measured at the physical layer. The Multipath Source Routing (MSR) protocol [12] is a multipath extension to DSR uses weighted round robin packet distribution to improve the delay and throughput. (Split Multipath Routing) [13] is another DSR extensions, which selects hop count limited and maximally disjoint multiple routes. Node-Disjoint Multipath Routing (NDMR) [14], provides with node-disjoint multiple paths.

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Other energy aware multipath protocols which give disjoint paths are Grid-based Energy Aware Node-Disjoint Multipath Routing Algorithm GEANDMRA) [15], Energy Aware Source Routing (EASR) [I6] and Energy Aware Node Disjoint multipath Routing (ENDMR)[I7]. The Lifetime-Aware Multipath Optimized Routing (LAMOR)[18] is based on the lifetime of a node which is related to its residual energy and current traffic conditions. Cost- effective Lifetime Prediction based Routing (CLPR) [19], combines cost efficient and lifetime predictions based routing. Minimum Transmission Power Routing (MTPR) [20], Power-aware Source Routing (PSR)[21].

2.1 Dynamic Source Routing Protocol (DSR)

The Dynamic Source Routing (DSR) [5] is a reactive unicast routing protocol that utilizes source routing algorithm. In source routing algorithm, each data packet contains complete routing information to reach its destination. In DSR each node also maintains route cache to maintain route information that it has learnt.

There are two major phases in DSR [5], the route discovery phase and the route maintenance phase. When a source node wants to send a packet, it firstly checks its route cache. If the required route is available, the source node includes the routing information inside the data packet before sending it. Otherwise, the source node initiates a route discovery operation by broadcasting route request packets. A route request packet contains addresses of both the source and the destination and a unique number to identify the request. Receiving a route request packet, a node checks its route cache. If the node doesn't have routing information for the requested destination, it appends its own address to the route record field of the route request packet. Then, the request packet is forwarded to its neighbors.

To limit the communication overhead of route request packets, a node processes route request packets that both it has not seen before and its address is not presented in the route record field. If the route request packet reaches the destination or an intermediate node has routing information to the destination, a route reply packet is generated. When the route reply packet is generated by the destination, it comprises addresses of nodes that have been traversed by the route request packet. Otherwise, the route reply packet comprises the addresses of nodes the route request packet has traversed concatenated with the route in the intermediate node's route cache.

III. Problem Issue

Nodes in Mobile Ad hoc Networks (MANETs) [1, 2] are battery driven. Thus, they suffer from limited energy level problems. Also the nodes in the network are moving, if a node moves out of the radio range of the other node, the link between them is broken. .Thus, in such an environment there are two major reasons of a link breakage:

a) Node dying of energy exhaustion

b) Node moving out of the radio range of its neighboring node

Hence, to achieve the route stability in MANETs, both link stability and node stability is essential.

The above mentioned techniques consider either of the two issues. Techniques in [19, 10, 13, and 20] calculate only multiple paths. Both stability issues are neglected in these. The work in [11] measures route quality in terms of SINR, which gives reliable links, but overall networks stability is not considered. Though [19] uses lifetime of a node as a generalized metric, it does not considers the mobility and energy issues which are critical to network - lifetime estimation. The protocol in [17] considers the energy issues in terms of the energy expenditure in data transmission, but the lifetime of the node and mobility factor is not discussed [7, 15, 16, 21] consider only energy metric to route the traffic.

Also, to send a packet from a source to destination many routes are possible. These routes can be either link disjoint or node-disjoint. Node disjoint protocols have an advantage that they prevent the fast energy drainage of a node which is the member of multiple link disjoint paths [14]. Hence, a technique which finds multiple node-disjoint paths considering both link and node stability has been proposed. The attempt is to find multiple node disjoint routes which consider both link stability and the node stability on their way.

IV. Metrics Used

To measure link and node stability together we are using two metrics, Link Expiration Time (LET) [19] and Energy Drain Rate (EDR) [22] respectively. These two metrics can be used to generate a composite metric which keeps track of the stability level of the entire path. .

Mobility Factor: The mobility factor Link Expiration Time (LET) was proposed in [19], by using the motion parameters (velocity, direction) of the nodes. It says that if r is the transmission distance between the two nodes, i and j, (xi, yi) and (xj, yj) be the position co-ordinates and (vi, θi) and (vj, θj) be the (velocity, direction) of motion of nodes. LET is defined as:

$$LET=-(ab+cd) +Q/(a^2+c^2) \qquad (1)$$

Where, $Q= \sqrt{ \{(a^2+c^2) r^2- (ad- bc)^2\} }$ and,

$a= v_i \, Cos\theta i - v_j \, Cos\theta_j$, $b= x_i-x_j$, $c= v_i \, Sin\theta i - vj \, Sin\theta_j$, and $d= y_i - y_j$

The motion parameters are exchanged among nodes at regular time intervals through GPS. .The above parameter suggests that if the two nodes have zero relative velocity, i,e, vi =vj and θj = θj, the link will remain forever, as, LET will be ∞.

Energy factor: Most of the energy based routing algorithms [10, 17, and 21], send large volume of data on the route with maximum energy levels, As a result, nodes with much higher current energy levels will be depleted of their battery power very early. The mobile node also loses some of it energy due to overhearing of the neighboring nodes. Thus, a node is losing its power over a period of time even if no data is being sent through it. Viewing all these factors a metric called Drain Rate (DR) was proposed in [22], Drain Rate of a node is defined as the rate of dissipation of energy of a node. Every node calculates its total energy consumption every T sec and estimates the DR, Actual Drain Rate is calculates by exponentially averaging the values of DR_{old}. and DR_{new} as follows:

$$DRi = \alpha DR_{old} + (1-\alpha)\, DR_{new} \quad (2)$$

Where, $0 < \alpha < l$, can be selected so as to give higher priority to updated information. Thus, higher the Drain Rate, faster the node is depleted of its energy.

V Proposed Work: Node Disjoint Multipath Routing Considering Link and Node Stability (NDMLNR)

The main aim of the proposed work is to find the multiple node disjoint routes from source to a given destination. The routes selected are such that all the links of the routes are highly stable. This will increase the lifetime of the route. Also it keeps track of the route bandwidth which can be further used by the source to select the optimal routes. From the factors Link Expiration Time (LET) [19] and Drain Rate (DR) [22] it is inferred that the Link Stability:

a) Depends directly on Mobility factor

b) Depends inversely on the energy factor

Hence, Link Stability Degree (LSD) is defined as:

LSD = Mobility factor / Energy factor (3)

It defines the degree of the stability of the link. Higher the value of LSD, higher is the stability of the link and greater is the duration of its existence. Thus, a route having all the links with LSD> LSD_{thr} is the feasible route.

We choose the Dynamic Source Routing (DSR) [5] protocol as a candidate protocol, details of which are given in section 2. Modifications are made to the Route Request (RREQ) and Route Reply (RREP) packets to enable the discovery of link stable node disjoint paths. The proposed scheme has three phases:

I. Route Discovery

2. Route Selection

3. Route Maintenance

The various phases are described as follows:

A Route Discovery

The source node when needs to send packet to some destination node, starts the route discovery procedure by sending the Route Request packet to all its neighbors .In this strategy , the source is not allowed to maintain route cache for a long time, as network conditions change very frequently in terms of position and energy levels of the nodes. Thus, when a nodes needs route to the destination, it initiates a Route Request packet, which is broadcasted to all the neighbors which satisfy the broadcasting condition.

The Route Request(RREQ) packet of the DSR [5] is extended as RREQ of the NDMLNR adding two extra fields, LSD and Bandwidth, B as shown in figure I. RREQ contains type field, source address field, destination field, unique identification number field, hop field, LSD, Bandwidth (cumulative bandwidth), Time -to-Live field and path field.

Type (T) field: It indicates the type of packet, SA (Source Address) field: It carries the source address of node. ID field: unique identification number generated by source to identify the packet. DA (Destination Address) field: It carries the destination address of node. Time to Live (TTL) field: It is used to limit the life time of packet, initially, by default it contains zero. Hop field: It carries the hop count; the value of hop count is incremented by one for each node through which packet passes. Initially, by default this field contains zero value. LSD field: when packet passes through a node, its LSD value with the node from which it has received this packet is updated in the LSD field. Initially, by default this field contains zero value. Bandwidth field carries the cumulative bandwidth of the links through which it passes; initially, by default this field contains zero value. Path field: It carries the path accumulations, when packet passes through a node; its address is appended at end of this field. The figure I. shows the RREQ packet.

SA	DA	Type	ID	TTL	Hops	Bandwidth	LSD	Path

Fig 1 RREQ packet

The Route Reply packet (RREP) of the DSR [5] is extended as RREP of the NDMLNR adding Bandwidth field. It is sent by the destination node after selecting the node disjoint paths among the various RREQ packets reaching it.

In DSR [5], when an intermediate node receives a RREQ packet, it checks whether its own address is already listed in the path list of received RREQ packet. If its address is not found, it appends its address to the route record of received RREQ and it is broadcasted to all its neighbors. Otherwise, the received RREQ packet will be dropped.

In the NDMLNR when an intermediate node receives a RREQ packet, it performs the following tasks:

I. Checks whether its own address is already listed in the route record of received RREQ packet. If its address is not found, it appends its address to the path list.

2. When an intermediate node receives a RREQ for the first time, it introduces a Wait Period, W. for the subsequent packets if any, with same identification number, traveling through different paths. It updates the value of LSD corresponding to the link on which it received the RREQ packet in the LSD field. It then checks its neighbors for QoS parameters, bandwidth here. Only those neighbors having LSD> LSD_{thr} and Link Bandwidth >= B are considered for broadcasting. Once the neighbors with required LSD are selected, node forwards packet. Later if an intermediate node receives duplicate RREQ packets with same (Source address and ID), as received from other paths, those duplicate RREQ packets will be dropped.

3. Every node maintains a Neighbor Information Table (NIT), to keep track of multiple RREQs. With following entries Source Address, Destination Address, Hops, LSD, ID and bandwidth.

SA	DA	ID	Hops	LSD	Bandwidth

Fig 2. Neighbor Information Table (NIT)As a RREQ reaches a node it enters its information in the NIT. It makes all the entries for the requests till Wait Period. At the end of

the Wait Period, it accepts the request with the highest value in LSD field. It adds the value of the link bandwidth to the Bandwidth field of the RREQ packet. If two RREQs have same LSD values, the one with lesser value of hop count isselected. In case, hops are also same, one with higher bandwidth is selected. In the worst case, RREQ is selected on First-come-first -serve basis. This prevents loops and unnecessary flooding of RREQ packets.

4. None of the intermediate nodes is allowed to send RREP if it has the current route to the destination. As doing this may lead to those paths which do not fulfill current QoS requirements.

In the NDMLNR, when the destination receives multiple RREQs it selects the paths with disjoint nodes. It then generates several replies and unicasts them to the source. Before that it appends its address and adds total bandwidth to each route request. Now each route reply that reaches the source contains a node-disjoint path from source to destination. Hence, source knows all the paths to the destination and their respective bandwidths. In case of two paths with one or more nodes common, the path with higher bandwidth is selected.

B. Route Selection

When the source node receives the RREPs from the multiple paths, it sorts the paths in the order of the increasing bandwidth. Depending on the bandwidth the source decides to use the single path, or all of the paths. In case of the multiple paths with same bandwidths, path with minimum number of hops is selected. If the paths conflict on the number of hops, the source node selects the path on First-come-First-Serve basis.

C. Route Maintenance

In case, LSD of a node falls below LSD_{thr}, it informs its predecessor node of the node failure by sending the NODEOFF message. Once a node receives such a message, it sends the ROUTEDISABLE message to the source node. Source can then reroute the packets to the backup routes. If no backup route exists, the source then starts the route discovery procedure again. We explain this technique with a suitable example in section 7.

VI. Traffic Distribution

The above discussed technique may result in many paths from a given source to a destination. To achieve fairness in traffic allocation based on energy and stability constraints, there is a need to select few optimal paths and divide traffic over them. To select optimal paths, we use Average Bandwidth of all the paths as the deciding factor. Let B1, B2, B3 ….Bn be the bandwidths of n disjoint paths. Thus, average bandwidth, B_{avg}, will be:

$$(B1 + B2 + B3 \ldots + Bn) / n \qquad (4)$$

The optimal paths are only those paths which have their respective bandwidths equal to or greater than Bavg. Through this, we attempt to achieve the stable and long lasting paths. Also, the paths are given load based on their capacity.

To divide the traffic among these optimal paths we use proportional distribution. If suppose, B_1, B_2 and B_3 are the bandwidths of the three selected optimal paths. Then B_1 gets $B_1/(B_1+B_2+B_3)$ percent of the total traffic, B_2 gets $B_2/(B_1+B_2+B_3)$ percent of the traffic and, so on .

For example, let there be three paths P_1, P_2 and P_3 with total bandwidths 20, 10, 15 Mbps respectively. Their Average bandwidth, B_{avg} , according to equation (4) is 15 Mbps. Thus, only paths P_1 and P_3 are optimal paths.

To distribute the traffic on these paths, P_1 gets 20/(20+15) = 57% of the traffic and P_3 gets 15/(20+15) = 43% of the traffic.

VII. Example

Let us illustrate our technique with the following example network shown in figure 3. Suppose node 1 is the source node and node 6 is the destination. Let LSDthr equals to 15. Let B equals to 5 mbps.

To send the packet, node 1 checks its neighbors (2.4.7) for their LSD value Out of these node 7 has value 9<15. So, node 1 sends the packets only to nodes 2 and 4.

Node 2 receives this packet for the first time, makes entry in its NIT for the RREQ packet as (1, 6, 1, 1, 20, 8) and starts Wait Time, 5 secs here. Node 2 now checks its neighbors, updates the path field as,1-2 and the bandwidth field to 8 and forwards RREQ to both 4 and 3. At node 4, it may receive two RREQ packets during Wait Time. One from node 1 directly, and, the other via node 2. It has two entries in its NIT (1,6,1,1.20,8) and (1,6,1,2,17,13). At this moment it selects the one from node 1 with higher LSD value, 20. It updates the path field of the RREQ packet as 1-4 and the bandwidth field to 7. It forwards the packet to both its neighbors, 5 and 8, with LSD values 16 and 18 respectively. Node 3 has only one neighbor, 6 which satisfies the LSD value and hence, it updates RREQ path field as 1-2-3 and the bandwidth field to 14 and forwards the packet to node 6. Node 6 now receives a path from source node 1. It appends its own ID to it. Thus, first path is 1-2-3-6 and bandwidth of this path is 17. Node 5 after receiving the RREQ packet with path 1-4, checks for its neighbors and forwards RREQ with updated path field to 1-4-5 and bandwidth field to14 to nodes 9 and 6 Node 6 now receives another path,1-4-5.It appends its ID to it, to get the path, 1-4-5-6 with bandwidth 19. Node 8 after receiving the RREQ packet forwards it to its neighbor, 9, after updating path field to 1-4-8 and bandwidth field to 15 Node 9 can receive two packets in its wait time, one from node 5 and the other from node 8. It updates its NIT as (1,6,1,3,16,22) and (1,6,1,3,18,21). To select from the one, it chooses one from node 8 as its LSD value is higher, 18. It then forwards the request after updating the path field as 1-4-8-9 and bandwidth field to 21. Node 6 again receives another path 1-4-8-9.It appends its ID to this path to get 1-4-8-9-6 with bandwidth 28.Now node 6 receives two paths 1-4-5-6 and 1-4-8-9-6 with node 4 as common node. It selects the one with higher bandwidth i.e. Path, 1-4-8-9-6 with bandwidth 28.

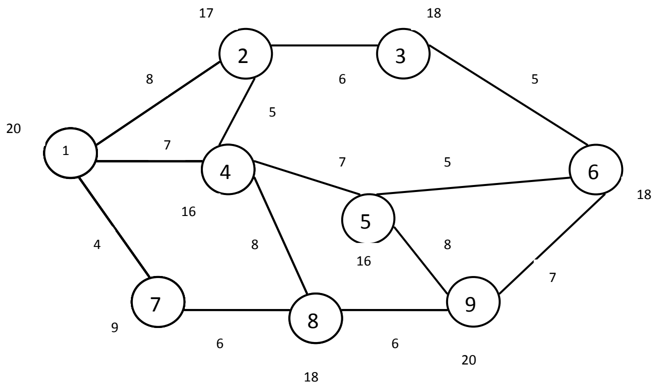

Fig 3. An example network.

VIII. Experimental Evaluation

simulate Node Disjoint Multipath Routing Considering Link And Node Stability (NDMLNR). We compare our NDMLNR protocol with the Energy Aware Node Disjoint Multi path Routing (ENDMR)[17] protocol In our simulation the channel capacity of mobile hosts is set to the same value: 11 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. Mobile nodes move in a 1500 meter x 300 meter rectangular region for 100 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 our simulation, the speed is set as 10m/s. The simulated traffic is Constant Bit Rate (CBR). The number of mobile nodes is varied as 10, 20,….50 and the pause time of the mobile node is varied as 10,20,30,40 and 50 seconds. Table 1. summarizes various simulation parameters.

Both NDMLNR and ENDMR use energy awareness; generate multiple paths that are node disjoint paths. The NDLMNR adds to the stability of ENDMR by considering the stability of the nodes and the links containing those nodes. The ENDMR protocol balances node energy utilization to increase the network lifetime. It takes network congestion into account to reduce the routing delay and increases the reliability of the packets reaching the destination.

The performance of the two protocols is compared using following metrics: Average Packet Delivery Ratio, throughput and average energy of the nodes.

In simulation we increase the number of nodes as 10, 20, 30….50.We study the performance of our protocol under this scenario. The graphs show the results for various metrics. Thus, the scenario presents the performance of the protocols under varying density of nodes

No. of Nodes	10,20,30,..50
Area Size	1500 X 300
Mac	802.11
Radio Range	250m
Simulation Time	100 sec

Traffic Source	CBR
Packet Size	512
Mobility Model	Random Way Point
Speed	10m/s
Pause time	10,20,30,….50
Initial energy	5.1 J
Sending power	0.660
Receiving power	0.395
Idle Power	0.035

Table 1 Simulation parameters

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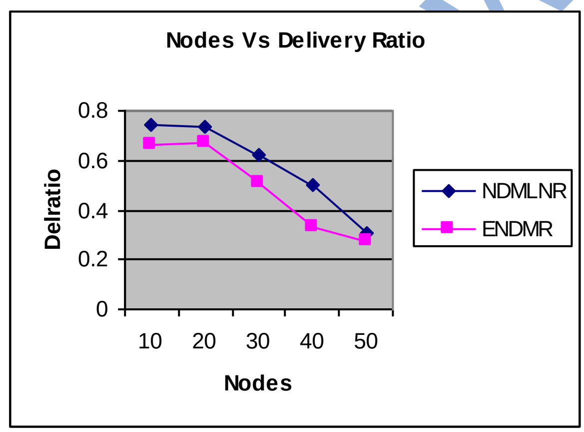

Fig 4. Nodes vs Delivery Ratio

The results from figure 4 show that considering the combined effect of energy and mobility factors, NDLMNR gives higher average packet delivery ration than ENDMR. Through this, it can be inferred that the paths found by NDLMNR are stable and have higher network lifetime as compared to ENDMR. ENDMR considers paths with nodes having the highest reaidual energy. In case, few nodes are not capable to comply with the needs and lifetime of traffic, they will die soon and hence, lower delivery ratio. Proportional distribution of load on the paths also leads to higher average delivery ratio.

The throughput of NDLMNR is higher as compared to ENDMR, as inferred from figure 5. This shows that selecting the paths considering the drain rate of nodes as energy parameter is more efficient than the residual energy of the nodes. Also, the higher throughput also accounts from the balanced traffic distribution on the node disjoint paths. The stability of both the nodes, from the drain rate and links, from link expiration time, results in the overall highly stable network and hence, higher throughput.

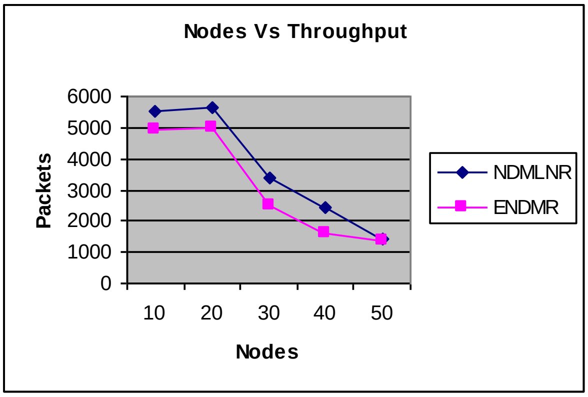

Fig5. Nodes vs Throughput

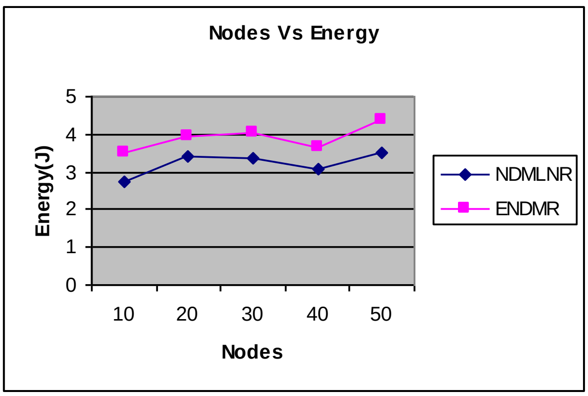

Fig 6. Nodes vs Energy consumed

The results in figure 6 clearly depict that the energy consumed by the nodes is lesser in NDLMNR as compared to ENDMR. This may be due to the selection of the nodes having higher stability and efficient distribution of traffic along the paths.Higher the stable nodes, higher is the path stability. High stability of the paths leads to lesser control packets needed for path maintenance and lesser energy consumption

IX Conclusion

The above mentioned technique considers the stability of the network from all aspects. The lifetime of the network can be reduced primarily by two causes. First, the node moving out of the radio range can lead to link breakage. Second, the node can be drained of its energy leading to network partitioning. The metric used in the proposed technique measures the stability of the network based on these two factors. The routing decisions at each node leads to the multiple paths, which are node disjoint. Doing this we attempt to prevent over usage of a single path nodes of which may drain out soon. Thus, this technique provides highly stable, reliable, robust node disjoint paths. As the paths are node disjoint, energy drain rate of the nodes is be less and hence longer lifetime. Also the paths are selected on the bandwidth constraints; they are the ones with higher capacity. The selected paths with higher bandwidth are further refined to select optimal paths having bandwidth higher than a threshold. This attempts to achieve stable and high capacity paths. The balancing of load on multiple paths also enhances the stability and lifetime of the networks and hence, higher throughput.Thus in this technique, as the routes are selected completely satisfying stability and capacity constraints, it fully complies with Quality of Service objectives.

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A Novel Routing Fusion algorithm for Topology aware Wireless Sensor Networks

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Abstract:**In this research paper, we consider wireless sensor networks in which the topology is under the control of the user. There are several applications in this category. We propose a novel levelling and clustering algorithm in such networks. By comparing the algorithm with earlier protocols, performance evaluation is carried out. It is realized that the algorithm can also be used for localization.**

I. Introduction

Wireless Sensor Networks is recognized as the ten technologies that will change the world. In the design of such networks various research problems needs to be solve. Some of them are routing, data fusion, localization etc. It is well recognized that these problems have solutions which depend on the application in mind. Broadly the application are classified into two groups

i. Those in which the topology of the network is under the control of the user.
ii. Those in which the topology is not under control of user.

Large number of protocols is designed for the application of wireless sensor networks in the category (ii). We realized that the applications in category (i) need routing, fusion, localization protocols which capitalize the fact that the topology is under the control of the user. An effort in this directions resulted in this research paper. This research paper is organised as follows. In section II, we summarize the previous research literature. In section III, we present our proposed algorithm. Section IV, we present performance evaluations. Section V concludes this paper.

II. Related work

In [5], the authors introduced a hierarchical clustering algorithm for sensor networks, called Low Energy Adaptive Clustering Hierarchy (LEACH). LEACH is a probabilistic distributed clustering algorithm. LEACH randomly selects a few sensor nodes as cluster heads (CHs) and rotates this role to evenly distribute the energy load among the sensors in the network. In [6], the authors introduced a Hybrid, Energy-Efficient, Distributed Clustering Approach (HEED), a clustering algorithm in which the initial probability for each node to become a tentative CH depends on its residual energy, and the final CHs are selected according to the cost. It is well known that flooding routing protocol is easy to implement but waste the network resources. Thus one is lead to "Gossiping" [4] as a better alternative. After careful understanding of variations of flooding protocol we proposed a directed flooding protocol which conserves the network resources. In [2], the authors proposed a levelled controlled Gossip technique which is used in Tsunami warning systems. In this paper, the authors partition the sensor field into circular levels of increasing radius from the base station (BS). Level controlled gossip is a technique that is being proposed which employs circular levelling and gossiping together. In [3], the authors proposed a novel localization algorithm using levelling (based on circular levels) and sectorization.

III. Issuses addressed and underlined assumptions

The following are the issues addressed in our proposed protocol:

- Reduce the numbers of nodes involved in the data transfer through virtual grid placement and levelling (rectangular levels).
- Avoids unnecessary flooding (i.e. discard data packets) throughout the network.
- Creates an energy efficient path towards the base station and hence is power aware.
- Increases the reliability of the gateways (routes) and reduces the complexity of the overall network.

The assumptions made on the nature of the sensor network are as follows:

- The nodes in the network are assumed to be stationary.
- The sensor network is densely deployed.
- Topology is under control of the user.
- All nodes are considered to have similar capabilities in the network.
- Base station (BS) is in the centre of the sensor field.

IV. Proposed algorithm

To decrease the consumption of energy of the battery, in our proposed algorithm instead of broadcasting the data packet into network, initially we are maintaining the data base of the adjacent cluster heads(CHs) as CH-table and only to selected CHs data packet is being forwarded.Our proposed algorithm has two phases *setup phase* and *routing phase*. In first phase total sensor field is divided into permanent grid structure using a global location information irrespective of the number of events. After the sensing field is divided into the grid structure, sensor nodes decide their grids (clusters) based on the location information and the CH is randomly selected which is responsible for aggregating the data in the cluster and forward it to sink. Each CH maintains a CH-

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table. From CH-table next-CH is selected and this process is explained in section 4.2.

When an event occurs, an event node sends a data packet to CH.CH already has the CH-table. It selects the CH with level number one lesser than current level number and column number one higher than the current column and row number of grid.

Setup phase

Setup phase has mainly three steps. They are as follows

a. Placing virtual grid and grid numbering

The sensor field is divided into a virtual grid using a global location information which is provided by localization system, such as GPS (Global Positioning System) or through techniques such as [1]. When sensor nodes are deployed in the field, they decide their grids with location information. The size of the grid is $\alpha \times \alpha$ and represented as ij where i represents row number and j represents column number

Now total sensing field is divided into I rows and J columns. Corner right side top most grid is numbered as00. Column number is incremented on moving left from grid 00 till the column with BS. Column next to BS station is numbered column number as $J-1$ and next onwards the column number is decremented on moving left. Row number is incremented on moving from top (00) to bottom till the row with BS. Row next to BS is numbered row as $I-1$ and next onward the row number is decremented on moving down. This is illustrated in fig. 1.

b. Clustering

Each grid is considered as a cluster. In order to select the head of each cluster, if sensor nodes begin with equal battery power, all sensor nodes locally flood a packet (head–packet) during the random period. The announcement of the head-packet is limited within the single cluster by simply dropping the packet from neighbour clusters. The sensor node which sends the head-packet first plays the role of CH.

c. Rectangular levelling

Rectangular levelling is done by using "hop count based method". In this method hop count is used to determine the levels. Initially hop count of CHs is set to infinity (or arbitrary large number). First BS broadcasts packets with hop count field set as zero. CHs which receive this packet set their hop count field as zero and level as one and the hop count field in the packet is incremented by one. These updated packets are broadcasted again. CHs that receive these packets update their level to *'hop count + 1'*, if their current level is higher than '*hop count + 1*'. If CHs that are having their level equal to or less than the '*hop count + 1*' value of the received packet, then their do not update their current level value.

This way the whole network is assigned as levels based on their hop count from the sink. Fig.1 depicts the sensing field at the end of setup phase.

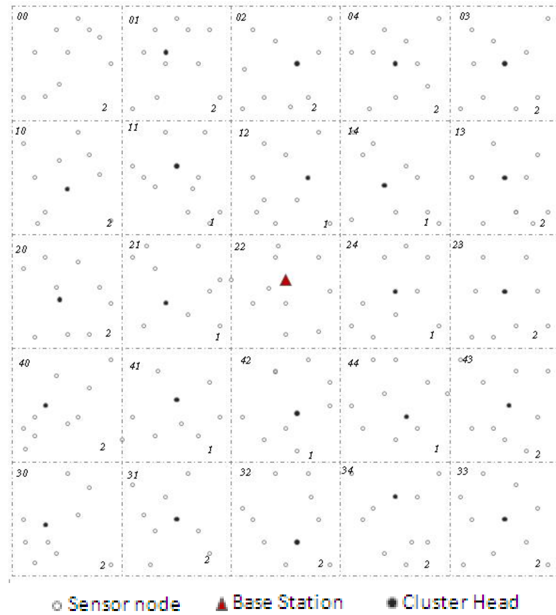

Fig. 1 sensor field at the end of setup phase, in each grid number in top right represents grid number and bottom left level number.

D . Routing phase

After the end of first phase CHs need to maintain adjacent CHs information in CH-table. So after the first phase each CH broadcasts information-packet. Information-packet consists of three fields, *level number, grid number and CH ID*. CHs which receive this packet update their CH-table with the information-packet. Being grid like structure each CH has only eight adjacent CHs.When an event occurs at any node say it as event node the packet it routed to BS as follows:

1. Event node sends the packet to CH.
2. CH sends the packet to *next-CH*. *Next-CH* is chosen from CH-table if and only if it follows following conditions
   i. *Level of next-CH must be one lesser than current CH.*
   ii. *Column or row number must be one greater than current grid column and row number respectively but not both column and rows.*
   iii. *If there is no such CH which satisfies above condition then select next-CH as the one with level lesser by one of current CH and column and row numbers greater by one of current grid column and row numbers respectively.*
3. Step 2 is followed until packet is send to BS.

This algorithm is illustrated in fig. 2.

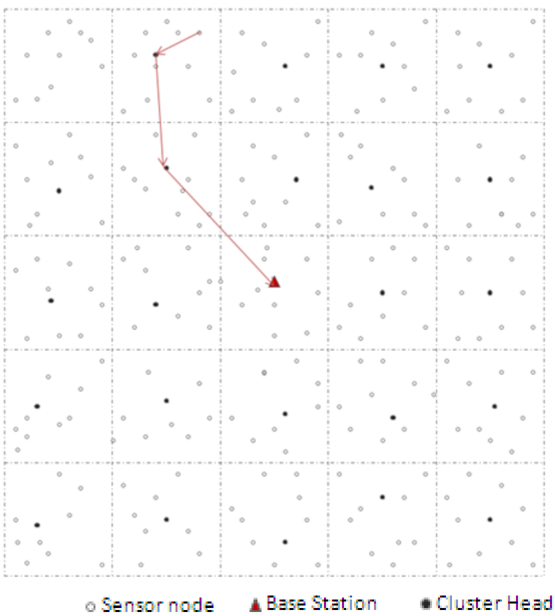

Fig. 2: routing of packet from event node to BS.

A. Cluster Rotation

Limited battery capacity of sensor nodes make CHs to drain out of power after a certain amount of time, which causes failure of transmission. To overcome this issue cluster rotation algorithm is implemented.
When CH remains with 5% of its total power it broadcasts a *CH-rotate* packet. This broadcast is limited to its cluster (grid) only. Nodes which receive this packet send *head-packet* to CH. Node with highest power is selected as CH.

V. PERFORMANCE EVALUATION

In this section, we have made comparative performance evaluation of our proposed algorithm with Flooding, Gossip and traditional circular levelling. The results have proven that the approach proposed by us increase the lifetime of the network and is thus the energy efficient. These results were plotted by several runs of the experiment.

A. Network Model

For the purpose of evaluating the algorithms, we simulated them by varying the number of nodes in the network. For each algorithm we started with a 100 node network and thereby generating the number of events that the network could handle. Similarly, the number of nodes has been varied up to 900 nodes and the corresponding number of events that these networks could handle was plotted in the plot. The two metrics of interest provided by the simulator are*Number of events:* It defines the life time of the network. *Number of Nodes*: The number of nodes that are present in the network.*Lifetime:* The time to failure of say 5% or 10% of nodes closer to the base station is considered to be the lifetime. The closeness to BS is measured based on the level number.

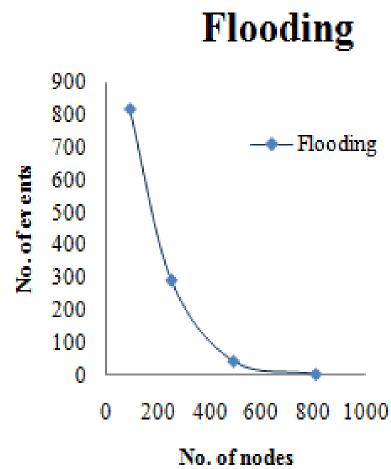

Fig.3: Nodes VS Network lifetime for flooding

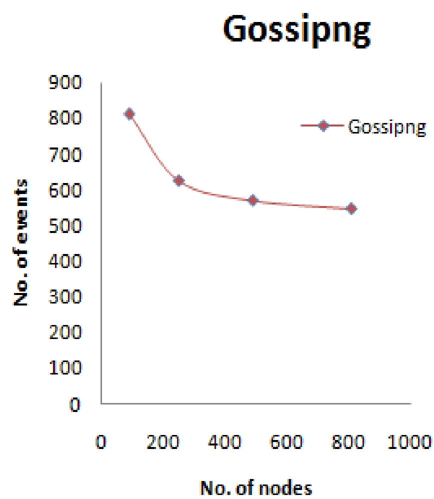

Fig. 4: Nodes VS Network lifetime for Gossip based approach.

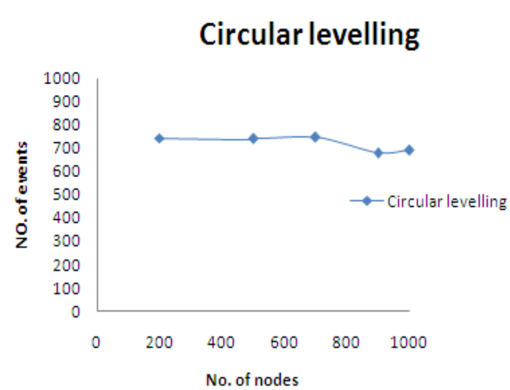

Fig. 5: Nodes VS Network lifetime for Circular levelling approach.

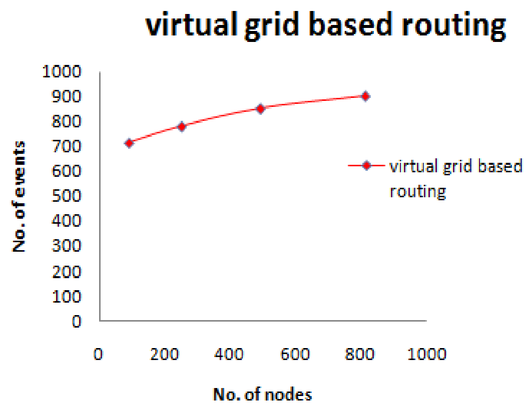

Fig. 6: Nodes VS Network lifetime for our proposed algorithm.

VI. Conclusions

In this paper we have proposed a routing and fusion algorithm for topology under control Wireless Sensor Networks We have made comparative evaluation with existing algorithms and this has proven that our proposed algorithm performs better than previous algorithms.

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A Parameter Based Modified Fuzzy Possibilistic C-Means Clustering Algorithm for Lung Image Segmentation

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Abstract*- Image processing is a technique necessary for modifying an image. The important part of image processing is Image segmentation. The identical medical images can be segmented manually. However the accurateness of image segmentation using the segmentation algorithms is more when compared with the manual segmentation. Medical image segmentation is an indispensable pace for the majority of subsequent image analysis tasks. In this paper, FCM and different extension of FCM Algorithm is discussed. The unique FCM algorithm yields better results for segmenting noise free images, but it fails to segment images degraded by noise, outliers and other imaging artifact. This paper presents an image segmentation approach using Modified Fuzzy Possibilistic C-Means algorithm (MFPCM). This approach is a generalized adaptation of standard Fuzzy C-Means Clustering (FCM) algorithm and Fuzzy Possibilistic C-Means algorithm. The drawback of the conventional FCM technique is eliminated in modifying the standard technique. The Modified FCM algorithm is formulated by modifying the distance measurement of the standard FCM algorithm to permit the labeling of a pixel to be influenced by other pixels and to restrain the noise effect during segmentation. Instead of having one term in the objective function, a second term is included, forcing the membership to be as high as possible without a maximum frontier restraint of one. Experiments are carried out on real images to examine the performance of the proposed modified Fuzzy Possibilistic FCM technique in segmenting the medical images. Standard FCM, Modified FCM, Possibilistic C-Means algorithm (PCM), Fuzzy Possibilistic C-Means algorithm (FPCM) and Modified FPCM are compared to explore the accuracy of our proposed approach.

Keywords-Fuzzy C-Means Clustering Algorithm ,Modified FCM, Modified Fuzzy Possibilistic C-Means Clustering Algorithm, Lung Nodule Detection, Medical Image Processing and Image Segmentation.

I. Introduction

Image segmentation is a necessary task for image understanding and analysis. A large variety of methods have been proposed in the literature. Image segmentation can be defined as a classification problem where each pixel is assigned to a precise class. Image segmentation is a significant process for successive image analysis tasks.

In general, a segmentation problem involves the division a given image into a number of homogeneous segments, such that the union of any two neighboring segments yields a heterogeneous segment. Numerous segmentation techniques have been proposed earlier in literature. Some of them are histogram based technique, edge based techniques, region based techniques, hybrid methods which combine both the edge based and region based methods together, and so on [1]. In recent years image segmentation has been extensively applied in medical field for diagnosing the diseases.

Image segmentation plays an important role in a variety of applications such as robot vision, object recognition, and medical imaging [2]. In the field of medical diagnosis an extensive diversity of imaging techniques is presently available, such as radiography, computed tomography (CT) and magnetic resonance imaging (MRI) [3, 4]. In recent times, Computed Tomography (CT) is the most effectively used for diagnostic imaging examination for chest diseases such as lung cancer, tuberculosis, pneumonia and pulmonary emphysema. The volume and the size of the medical images are progressively increasing day by day. Therefore it becomes necessary to use computers in facilitating the processing and analyzing of those medical images. Even though the original FCM algorithm yields good results for segmenting noise free images, it fails to segment images corrupted by noise, outliers and other imaging artifact.

Medical image segmentation is an indispensable step for most successive image analysis tasks. This paper presents an image segmentation approach using Modified Fuzzy C-Means (FCM) and Fuzzy Possibilistic C-Means (FPCM) algorithm. Recently, many researchers have brought forward new methods to improve the FCM algorithm [5, 6]. This approach is a generalized version of standard Fuzzy C-Means Clustering (FCM) algorithm. The limitation of the conventional FCM technique is eliminated in modifying the standard technique. The algorithm is formulated by modifying the distance measurement of the standard FCM algorithm to permit the labeling of a pixel to be influenced by other pixels and to restrain the noise effect during segmentation. Possibilistic C-Means (PCM) algorithm, interprets clustering as a Possibilistic partition. Instead of having one term in the objective function, a second term is included, forcing the membership to be as high as possible without a maximum limit constraint of one. Experiments are conducted on real images to investigate the performance of the proposed modified FPCM technique in segmenting the medical images. Standard FCM, Modified FCM, Fuzzy

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Possibilistic C-Means and MFPCM algorithm are compared to explore the accuracy of our proposed approach.

The remainder of the paper is organized as follows. Section 2 provides an overview on related research works in medical image segmentation. Section 3 explains initially explains the standard FCM algorithm and latter it explains the proposed MFCM, FPCM and MFPCM algorithm. Section 4 discusses on experimental results for real images. Section 5 concludes the paper with fewer discussions.

Related work

A lot of research work has been carried on various techniques for image segmentation. In recent years, many researchers have brought forward new methods to improve the FCM algorithm [5, 6]. This section of the paper provides an overview on the related research work conducted on medical image processing.

Kenji Suzuki et al. in [7] presented an image processing technique using Massive Training Artificial Neural Networks (MTANN). Their approach resolve the problem faced by radiologists as well as computer-aided diagnostic (CAD) schemes to detect these nodules in case when the lung nodules overlaps with the ribs or clavicles in chest radiographs. An MTANN is extremely a non-linear filter that can be trained by use of input chest radiographs and the equivalent "teaching" images. They used a linear-output back-propagation (BP) algorithm that was derived for the linear-output multilayer ANN model in order to train the MTANN. The dual-energy subtraction is a technique used in [7] for separating bones from soft tissues in chest radiographs by using the energy dependence of the x-ray attenuation by different materials.

A robust statistical estimation and verification framework was proposed by Kazunori Okada et al. in [8] for characterizing the ellipsoidal geometrical structure of pulmonary nodules in the Multi-slice X-ray computed tomography (CT) images. They proposed a multi-scale joint segmentation and model fitting solution which extends the robust mean shift-based analysis to the linear scale-space theory. A quasi-real-time three-dimensional nodule characterization system is developed using this framework and validated with two clinical data sets of thin-section chest CT images. Their proposed framework is a combination of three different but successive stages. They are model estimation, model verification and volumetric measurements. The main issue of the approach is a bias due to the ellipsoidal approximation.

Segmentation-by-registration scheme was put forth by Ingrid Sluimer et al. in [9]. In the scheme a scan with normal lungs is elastically registered to a scan containing pathology. Segmentation-by-registration scheme make use of an elastic registration of inclusive scans using mutual information as a similarity measure. They are compared the performance of four segmentation algorithms namely Refined Segmentation-by-Registration, Segmentation by Rule-Based Region growing, Segmentation by Interactive Region growing, and Segmentation by Voxel Classification. The comparison results revealed that refined registration scheme enjoys the additional benefit since it is independent of a pathological (hand-segmented) training data.

A genetic algorithm for segmentation of medical images was proposed by Ghosh et al. in [10]. In their paper, they presented a genetic algorithm for automating the segmentation of the prostate on two-dimensional slices of pelvic computed tomography (CT) images. In their approach the segmenting curve is represented using a level set function, which is evolved using a genetic algorithm (GA). Shape and textural priors derived from manually segmented images are used to constrain the evolution of the segmenting curve over successive generations. They reviewed some of the existing medical image segmentation techniques. They also compared the results of their algorithm with those of a simple texture extraction algorithm (Laws texture measures) as well as with another GA-based segmentation tool called GENIE. Their preliminary tests on a small population of segmenting contours show promise by converging on the prostate region. They expected that further improvements can be achieved by incorporating spatial relationships between anatomical landmarks, and extending the methodology to three dimensions.

A novel approach for lung nodule detection was described by M. Antonelli et al. in [11]. They described a computer-aided diagnosis (CAD) system for automated detection of pulmonary nodules in computed-tomography (CT) images. Combinations of image processing techniques are used for extraction of pulmonary parenchyma. A region growing method based on 3D geometric features is applied to detect nodules after the extraction of pulmonary parenchyma. Experimental results show, that implementation of this nodule detection method, detects all malignant nodules effectively and a very low false-positive detection rate was achieved.

Xujiong Ye et al. in [12] presented a new computer tomography (CT) lung nodule computer-aided detection (CAD) method. The method can be implemented for detecting both solid nodules and ground-glass opacity (GGO) nodules. Foremost step of the method is to segment the lung region from the CT data using a fuzzy thresholding technique. The next step is the calculation of the volumetric shape index map and the "dot" map. The former mentioned map is based on local Gaussian and mean curvatures, and the later one is based on the Eigen values of a Hessian matrix. They are calculated for each Voxel within the lungs to enhance objects of a specific shape with high spherical elements. The combination of the shape index and "dot" features provides a good structure descriptor for the initial nodule candidate generation. Certain advantages like high detection rate, fast computation, and applicability to different imaging conditions and nodule types make the method more reliable for clinical applications.

A robust medical image segmentation algorithm was put forth by Wang et al. in [13]. Automated segmentation of images has been considered an important intermediate processing task to extract semantic meaning from pixels. In general, the fuzzy c-means approach (FCM) is highly effective for image segmentation. But for the conventional FCM image segmentation algorithm, cluster assignment is based exclusively on the distribution of pixel attributes in the feature space, and the spatial distribution of pixels in an image is not taken into consideration. In their paper, they presented a novel FCM image segmentation scheme by utilizing local contextual information and the high inter-pixel correlation inherent. Firstly, a local spatial similarity measure model is established, and the initial clustering center and initial membership are determined adaptively based on local spatial similarity measure model. Secondly, the fuzzy membership function is modified according to the high inter-pixel correlation inherent. Finally, the image is segmented by using the modified FCM algorithm. Experimental results showed the proposed method achieves competitive segmentation results compared to other FCM-based methods, and is in general faster.

II. Proposed Approach

A. Conventional Fuzzy C-Means Algorithm

Fuzzy C-Means (FCM) Clustering algorithm is one of the accepted approaches for assigning multi-subset membership values to pixels for either segmentation or other type of image processing [14]. Generally, FCM algorithm proceeds by iterating the two indispensable conditions until a solution is reached. Each data point will be joined with a membership value for each class after FCM clustering. The objective of FCM is to determine the cluster centers and to produce the class membership matrix. In other words, it assigns a class membership to a data point, depending on the similarity of the data point to a scrupulous class relative to all other classes. The class membership matrix is a cXN matrix; in which c is the number of groups and N is the number of samples. Let X={x1,……xn} be the training set and c≥2 be an integer. A fuzzy c-partition of 'X' can be represented by a matrix, U = {μik}∊RcXN . U can be used to describe the cluster structure of X, by evaluating μik, as a degree of membership of xk to cluster i. The codebook vectors are evaluated by minimizing the distortion measure given by the following equation,

$$\text{Minimize: } J_m(U, v) = \sum_{k=1}^{N} \sum_{i=1}^{c} (\mu_{ki})^m \|X_k - v_i\|^2 A$$

where $X=\{x_1, x_2, \ldots x_N\} \subset R^N$ in a dataset, c is the number of clusters in X: $2 \le c < N$, m is a weighting exponent: $1 \le m < \infty$, U = $\{\mu_{ik}\}$ is the fuzzy c-partition of X, $\|X_k - v_i\|\ A$ is an induced a-norm of R^N, and A is a positive-definite (NXN) weight matrix.

A conventional FCM algorithm includes the following steps,

1. Initially values are set for the parameters like c, A, m, ε, and the loop counter 't' is set to 1,
2. As a next step it is necessary to create a random cXN membership matrix U,
3. The cluster centers are then evaluated using the following equation,

$$v_i^{(t)} = \frac{\sum_{k=1}^{N} (\mu_{ki}^{(t)})^m X_k}{\sum_{k=1}^{N} (\mu_{ki}^{(t)})^m}$$

4. The membership matrix is updated periodically with the help of the following equation,

$$\mu_{ki}^{(t+1)} = \left[\sum_{j=1}^{c} \left(\frac{d_{ki}}{d_{kj}} \right)^{\frac{2}{m-1}} \right]^{-1}$$

Where d_{ki} is given by $\left\| X_k - v_i^{(t)} \right\| A$

5. If max $\left\{ \left| \mu_{ki}^{(t)} - \mu_{ki}^{(t-1)} \right| \right\}$ > ε, increment't' and go to step 3.

B. Modified Fuzzy C-Means Clustering Technique for image segmentation

The most important shortcoming of standard FCM algorithm is that the objective function does not think about the spatial dependence therefore it deal with image as the same as separate points. In order to decrease the noise effect during image segmentation, the proposed method incorporates both the local spatial context and the non-local information into the standard FCM cluster algorithm using a novel dissimilarity index in place of the usual distance metric. Therefore a modified FCM algorithm is used to segment the image in our proposed paper. The non-local means algorithm [15] [16] tries to take advantage of the high degree of redundancy in an image. The membership value decides the segmentation results and hence the membership value is evaluated by the distance measurement denoted as d_{ki}. Therefore the approach modifies the distance measurement parameter which is readily influenced by local and non-local information.

$$d_{ki}(x_j, v_i) = (1-\lambda_j)\ d_l^2(x_j, v_i) + \lambda_j\, d_{nl}^2(x_j, v_i)$$

where d_l stands for the distance measurement influenced by local information, and d_{nl} stands for the distance measurement influenced by non-local information, λ_j with the range from zero to one, is the weighting factor controlling the tradeoff between them.

The distance measurement influenced by the local measurement d_l is given by,

$$d_l^2(x_j, v_i) = \frac{\sum_{x_k \in N_j} \omega_l(x_k, x_j)\, d^2(x_k, v_i)}{\sum_{x_k \in N_i} \omega_l(x_k, x_j)}$$

Where $d^2(x_j, v_i)$ is the Euclidean distance measurement, $\omega_l(x_k, x_j)$ is the weight of each pixel in N_i.

The distance measurement influenced by non-local information d_{nl} is computed as a weighted average of all the pixels in the given image I,

$$d_{\mathrm{nl}}^2(x_j, v_i) = \sum_{x_k \in I} \omega_{\mathrm{nl}}(x_k, x_j)\, d^2(x_k, v_i)$$

Modified FCM algorithm goes through the following steps,

1. Set the number of clusters 'c' and the index of fuzziness 'm.' Also initialize the fuzzy cluster Centroid vector 'v' randomly and set $\varepsilon > 0$ to a small value,
2. Set the neighborhood size and the window size includes the evaluation of cluster centers and membership matrix,
3. Evaluate the modified distance measurement using the equation mentioned as $d_{ki}(x_j, v_i)$,
4. Update the membership matrix and the distance measurement.

C. Possibilistic C-Means Algorithm (PCM)

In order to overcome the limitations of conventional FCM technique, Possibilistic C-Means (PCM) has been proposed in this paper for medical image segmentation. The Possibilistic C-Means algorithm uses a Possibilistic type of membership function to illustrate the degree of belonging. It is advantageous that the memberships for representative feature points be as high as possible and unrepresentative points have low membership. The intention function, which satisfies the requirements, is formulated as follows,

$$\min\left\{J_m(x,\mu,c) = \sum_{i=1}^{c}\sum_{j=1}^{N}\mu_{ij}^m d_{ij}^2 + \sum_{i=1}^{c}\eta_i\sum_{j=1}^{N}(1-\mu_{ij})^m\right\}$$

where, dij represents the distance between the jth data and the ith cluster center, μij denotes the degree of belonging, m represents the degree of fuzziness, ηi is the suitable positive number, c is the number of clusters, and N denotes the number of pixels. μij can be obtained using the following equation,

$$\mu_{ij} = \frac{1}{1+\left(\frac{d_{ij}^2}{\eta_i}\right)^{\frac{1}{m-1}}}$$

The value of η_i determines the distance at which the membership values of a point in a cluster becomes 0.5. The main advantage of this PCM technique is that the value of η_i can be fixed or can be changed in each iteration. This can be accomplished by changing the values of d_{ij} and μ_{ij}. The PCM is more robust in the presence of noise, in finding valid clusters, and in giving a robust estimate of the centers.
Updating the membership values depends on the distance measurements [17].The Euclidean and Mahalanobis distance are two common ones. The Euclidean distance works well when a data set is compact or isolated [18] and Mahalanobis distance takes into account the correlation in the data by using the inverse of the variance-covariance matrix of data set which could be defined as follows,

$$D = \sum_{i,j=1}^{i,j=p} A_{ij}(x_i - y_i)(x_j - y_j)$$

$$A_{ij} = \rho_{ij}\sigma_i\sigma_j$$

where, x_i and y_i are the mean values of two different sets of parameters, X and Y. σ_i^2 are the respective variances, and ρ_{ij} is the coefficient of correlation between i^{th} and j^{th} variants.

D. Fuzzy Possibilistic C-Means Algorithm (FPCM)

FPCM algorithm was proposed by N.R.Pal, K.Pal, and J.C.Bezdek[18] and it includes both possibility and membership values. FPCM model can be seen as below:

$$\underbrace{\min}_{(U,T,V)} \{J_{m,\eta}(U,T,V;X)\} = \sum_{i=1}\sum_{k=1}(u_{ik}^m + t_{ik}^\eta)D_{ikA}^2$$

subject to the constraints

$$m > 1, \eta > 1, 0 \le u_{ik}, t_{ik} \le 1,$$

$$D_{ikA} = \|x_k - v_i\|_A,$$

$$\sum_{i=1}^{c} u_{ik} = 1 \forall k, i.e., U \in M_{fcn}$$

$$\sum_{k=1}^{n} t_{ik} = 1 \forall i, i.e., T^t \in M_{fnc}.$$

where U is membership matrix, T is possibilistic matrix, and V is the resultant cluster centers, c and n are cluster number and data point number respectively. The first order necessary conditions for extreme of $J_{m,\eta}$ are: If $D_{ikA} = \|x_k - v_i\|_A > 0$ for all i and k,m,η > 1 and X contains at least c distinct data points, then $(U, T^t, V) \in M_{fcn} \times M_{fcn} \times \mathbb{R}^p$ may minimize $J_{m,\eta}$ only if

$$u_{ik} = \left(\sum_{j=1}^{c}\left(\frac{D_{ikA}}{D_{jkA}}\right)^{2/(m-1)}\right)^{-1}$$

$$1 \le i \le c; 1 \le k \le n$$

$$t_{ik} = \left(\sum_{j=1}^{n}\left(\frac{D_{ikA}}{D_{ijA}}\right)^{2/(\eta-1)}\right)^{-1}$$

$$1 \le i \le c; 1 \le k \le n$$

$$v_i = \frac{\sum_{k=1}^{n}(u_{ik}^m + t_{ik}^\eta)x_k}{\sum_{k=1}^{n}(u_{ik}^m + t_{ik}^\eta)}, 1 \le i \le c.$$

The above equations show that membership u_{ik} is affected by all c cluster centers, while possibility t_{ik} is affected only by the i-th cluster center c_i. The possibilistic term distributes the t_{ik} with respect to all n data points, but not with respect to all c clusters. So, membership can be called relative typicality, it measures the degree to which a point belongs to one cluster relative to other clusters and is used to crisply label a data point. And possibility can be viewed as absolute typicality, it measures the degree to which a point belongs to one cluster relative to all other data points, it can reduce the effect of outliers. Combining both membership and possibility can lead to better clustering result.

E. Parameter based Modified Fuzzy Possibilistic Clustering Algorithm (MFPCM)

The clustering optimization is entirely dependent on objective function since the choice of a suitable objective function is the key to the success of the cluster analysis and to obtain better cluster results [20]. The following set of requirements is taken into account in order to get hold of a suitable objective function. The distance between the clusters must be maximized and in the same manner the distance between the clusters and the data points assigned to them should be minimized. The attraction between the data and the clusters is governed by the following objective function formula.

$$J_{FPCM}(U,T,V) = \sum_{i=1}^{c}\sum_{j=1}^{n}(u_{ij}^m + t^\eta)d?\,\mathbf{x}_j, v_i)$$

Where U is membership matrix, T is Possibilistic matrix, and V is the resultant cluster centers, c and n are cluster number and data point number respectively. The learning procedure of α is based on an exponential partition strength stuck between clusters and is updated at each iteration. The formula of this parameter is:

$$\alpha = \exp\left(-\min_{i \ne k}\frac{\|v_i - v_k\|^2}{\beta}\right)$$

β is chosen as a sample variable and it is a normalized term. Therefore β can be defined as follows:

$$\beta = \frac{\sum_{j=1}^{n}\|x_j - \bar{x}\|^2}{n} \quad \text{where } \bar{x} = \frac{\sum_{j=n}^{n} x_j}{n}.$$

This section proposes a new parameter that suppresses the common value of α and thereby replacing it by a new parameter similar to weight to each vector. The weight is evaluated as follows:

$$w_{ji} = \exp\left(-\frac{\|x_j - v_i\|^2}{\left(\sum_{j=1}^{n}\|x_j - \bar{v}\|^2\right) * c / n}\right)$$

where w_{ij} is weight of the point j in relation to the class i. this weight is used to adapt the fuzzy and typical partition. The objective function is poised of two expressions: the first is the fuzzy function and uses a fuzziness weighting exponent, the second is Possibilistic function and uses a typical weighting exponent. The objective function of the MFPCM can be formulated as follows

$$J_{MFPCM} = \sum_{i=1}^{c}\sum_{j=1}^{n}(\mu_{ij}^m w_{ji}{}^m d^{2m}(x_j, v) + t_{ij}^\eta w_{ji}{}^\eta d^{2\eta}(x_j, v_i))$$

U = {μ_{ij}} represents a fuzzy partition matrix, is defined as:

$$u_{ij} = \left[\sum_{k=1}^{c}\left(\frac{d?\,\mathbf{x}_j, v_i)}{d?\,\mathbf{x}_j, v_k)}\right)^{2m/(m-1)}\right]^{-1}$$

T = {t_{ij}} which resembles a distinctive partition matrix, is defined as:

$$t_{ij} = \left[\sum_{k=1}^{n}\left(\frac{d?\,\mathbf{x}_j, v_i)}{d?\,\mathbf{x}_j, v_k)}\right)^{2\eta/(\eta-1)}\right]^{-1}$$

V = {vi} represents c centers of the clusters, is defined as:

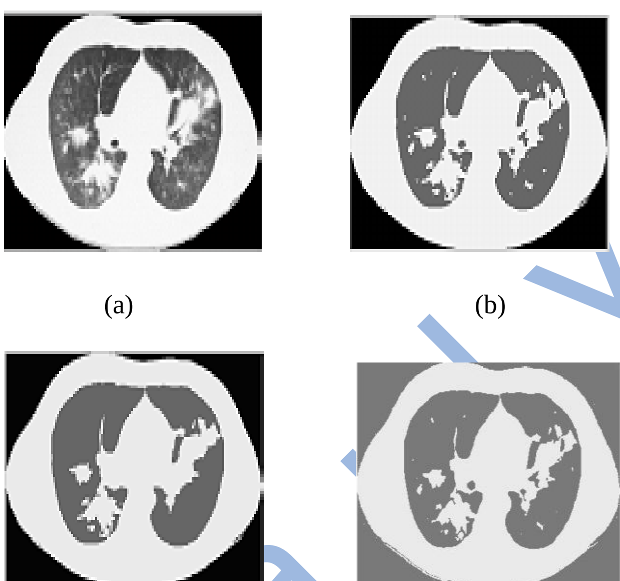
$$v_i = \frac{\sum_{j=1}^{n}(\mu_{ij}^m w_{ji}^m + t_{ij}^\eta w_{ji}^\eta) * x_j}{\sum_{j=1}^{n}(\mu_{ij}^m w_{ji}^m + t_{ij}^\eta w_{ji}^\eta)}$$

III. EXPERIMENTAL RESULTS

The proposed Modified FCM algorithm, Fuzzy Possibilistic C-Means and MFPCM algorithm is implemented using MATLAB and tested on real images to explore the segmentation accuracy of the proposed approach. The various types of FCM techniques that has been used are standard FCM, Modified FCM, PCM, FPCM and MFPCM Clustering algorithm and are compared.

A. *Real Image Dataset*

A real set of lung images are used to evaluate the accuracy of the proposed algorithm in segmenting the medical images. The results obtained are then compared with the segmentation results that were performed manually to explore the accuracy of the proposed algorithm. The segmentation results of standard FCM, Modified FCM, Possibilistic C-Means Clustering (PCM), Fuzzy Possibilistic C-Means Clustering (FPCM) and Modified Fuzzy Possibilistic C-Means Clustering (MFPCM) are considered to investigate the best algorithm that delivered better segmentation results for real medical images. The three most important parameters used to determine the accuracy of the proposed algorithm are similarity, false positive and the false negative ratio. From the results obtained it can be concluded that our proposed algorithm performed well ahead of other techniques in segmenting the real medical images. The three main attributes mentioned above i.e. similarity, false positive ratio, and the false negative ratios are listed in Table 1, for all the image segmentation techniques. Figure 1 shows the segmentation result of the different techniques



(a) (b)

(c) (d)

Figure 1 (a) Actual Image, Segmented Images (b) using Standard FCM (c) using FPCM (d) using MFPCM

Algorithm	Similarity	False Positive Ratio	False Negative Ratio
Standard FCM	86.03	20.15	8.50
Modified FCM	89.50	16.50	5.30
Possibilistic C-Means Clustering Algorithm (PCM)	91.00	14.50	4.70
Fuzzy Possibilistic C-Means Clustering (FPCM)	92.50	12.80	3.40
Modified Fuzzy Possibilistic C-Means Clustering (MFPCM)	94.25	10.60	2.80

Table 1 Different Indices for Different Algorithms

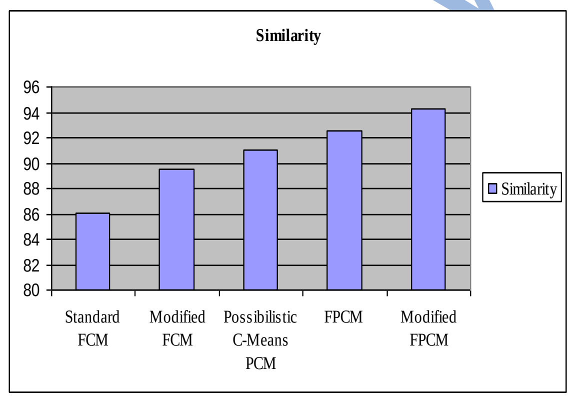


Figure 2 Comparison of Similarity

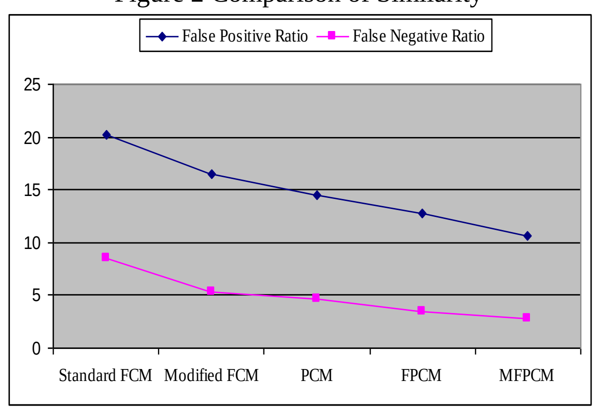


Figure 3 Comparison of False Positive and False Negative Ratio for the three approaches

The experimental results obtained by employing the Modified Fuzzy Possibilistic C-Means (MFPCM) algorithm revealed that the proposed technique of image segmentation has a better performance over other FCM extension methods. Furthermore, the proposed approach of image segmentation using Modified Fuzzy Possibilistic C-Means algorithm eliminates the effect of noise greatly. This in turn increased the segmentation accuracy of the proposed image segmentation technique.

IV. CONCLUSION

FCM is one of a conventional clustering method and has been generally applied for medical image segmentation. On the other hand, conventional FCM at all times suffers from noise in the images. Even though the unique FCM algorithm yields good results for segmenting noise free images, it fails to segment images corrupted by noise, outliers and other imaging artifact. Though a lot of researchers have developed a diversity of extended algorithms based on FCM, not any of them are ideal. A modified FCM clustering algorithm and Modified Fuzzy Possibilistic C-Means (MFPCM) algorithm is proposed in this paper. In the proposed Modified FCM algorithm, both local and non-local information are integrated to control the tradeoff between them. The algorithm is put together by modifying the distance measurement of the standard FCM algorithm to authorize the labeling of a pixel to be influenced by other pixels and to hold back the noise effect during segmentation. The Modified Fuzzy Possibilistic C-Means (MFPCM) algorithm interprets clustering as a Possibilistic partition and includes membership functions. Experiments are conducted on real medical images to estimate the performance of the proposed algorithm. The three most important parameters used to determine the accuracy of the proposed algorithm are similarity, false positive and the false negative ratio. The experimental results suggested that the proposed algorithm performed well than other FCM extension, segmentation algorithms

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Power Optimization In Mobile Ad Hoc Network

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***Abstract*- A mobile ad hoc network is a collection of mobile nodes that are dynamically and arbitrarily located in such a manner that the interconnections between nodes are capable of changing on a continual basis. Mobile ad-hoc networks are the autonomous systems of mobile nodes forming network in the absence of any centralized support. This is a new form of network and might be able to provide services at places where it is not possible otherwise. Absence of fixed infrastructure poses several types of challenges for this type of networking. Almost devices in mobile Ad hoc network run on lithium-ion rechargeable batteries, these batteries have a lifetime of few hours of active lifetime. To solve this problem researcher tried to optimized power consumption in every aspect of mobile devices. Power consumption can be reduced at device level, at transmission level or may be by using optimized power aware routing protocol. In this paper we have given a brief description of basic aspects of mobile ad hoc network and studied various power saving techniques in mobile ad hoc network & given a comparative analysis of these techniques.**

Keywords: Mobile Ad-hoc network, Transmission power, Routing protocol, Power saving.

I. Introduction

A mobile *ad hoc* network (MANET) is a network formed without any central administration, which consists of mobile nodes that use a wireless interface to send & receive packet data. Since the nodes in a network of this kind can serve as routers and hosts, they can forward packets on behalf of other nodes and run user applications. Ad- hoc networks [8] are formed where there is no existing infrastructure and there is a need for communication. Examples of ad hoc networks include soldiers on enemy terrain, workers in a disaster area, or a group of executives at an outdoor location. Figure 1 shows a typical ad hoc network.

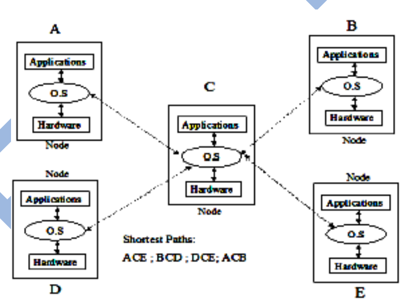

Figure 1. Ad hoc networks

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What differentiates ad hoc networks from traditional wireless networks is the absence of a centralized base station. In traditional wireless networks, nodes wishing to communicate with each other have to first contact the nearest base station, which forwards their requests to the base station closest to the destination node. All packets are routed through the path established by the base station. The base stations perform the tasks of tracking, routing and route maintenance. In ad hoc networks, all these tasks are performed by the nodes themselves, in addition to their personal tasks. This causes additional drain on the batteries leading to a diminished lifetime. Power utilization can be optimized by employing routing algorithms that avoid nodes with less battery power remaining while trying to minimize the total power consumed in transmitting a packet.

II. Manet Evolution

The whole life-cycle of ad-hoc networks could be categorized into three generations.
The first generation goes back to 1972. At the time, they were called PRNET (Packet Radio Networks). The PRNET used a combination of ALOHA[5] (Areal Locations of Hazardous Atmospheres) and CSMA (Carrier Sense Medium Access), approaches for medium access, and a kind of distance-vector routing. PRNET were used on a trial basis to provide different networking capabilities in a combat environment.

The second generation of ad-hoc networks emerged in 1980s, when the ad-hoc network systems were further enhanced and implemented as a part of the SURAN (Survivable Adaptive Radio Networks) program. This provided a packet-switched network to the mobile battlefield in an environment without infrastructure. This program proved to be beneficial in improving the radios performance by making them smaller, cheaper, and resilient to electronic attacks.

The third generation of ad-hoc networks emerged in the 1990s, when the concept of commercial ad-hoc networks arrived with notebook computers and other viable communications equipment. At the same time, the idea of a collection of mobile nodes was proposed at several research conferences. In the meanwhile the IEEE 802.11 subcommittee had adopted the term "ad-hoc networks" and the research community had started to look into the possibility of deploying ad-hoc networks in other areas of application.

III. Features of Mobile Ad Hoc Network

In general, mobile ad hoc networks are formed dynamically by an autonomous system of mobile nodes that are connected via wireless links without using the existing network infrastructure or centralized administration. The nodes are free to move randomly and organize themselves arbitrarily; thus, the network's wireless topology may change rapidly and unpredictably. Such a network may operate in a standalone fashion, or may be connected to the larger network. Mobile ad hoc networks are infrastructure-less networks since they do not require any fixed infrastructure, such as a base station, for their operation.

Mobile Ad hoc network has the following features:

A. Autonomous and infrastructure-less

MANET does not depend on any established infrastructure or centralized administration. Each node operates in distributed peer-to-peer mode, acts as an independent router and generates independent data. Network management has to be distributed across different nodes, which brings added difficulty in fault detection and management.

B. Distributed operation

As there is no background network available for the central control of the network operations, the control and management of the network is distributed among the terminals. The nodes involved in a mobile ad hoc network should collaborate amongst themselves and each node acts as a relay as needed, to implement functions e.g. security and routing.

C. Multi-hop routing

In these kind of networks no default router available, every node acts as a router and forwards each others' packets to enable information sharing between mobile hosts.

D. Dynamic network topology

Since the nodes are mobile, the network topology may change rapidly and unpredictably and the connectivity among the terminals may vary with time. The mobile ad hoc network should adapt to the traffic and propagation conditions as well as the mobility patterns of the nodes. The mobile nodes in the network dynamically establish routing among themselves as they move about forming their own network on fly.

E. Variation in link and node capabilities

Each node may be equipped with one or more radio interfaces that have varying transmission/receiving capabilities and operate across different frequency bands [1, 2]. This heterogeneity in node radio capabilities can result in possibly asymmetric links. In addition, each mobile node might have a different software/hardware configuration, resulting in variability in processing capabilities. Designing network protocols and algorithms for this heterogeneous network can be complex, requiring dynamic adaptation to the changing conditions (power and channel conditions, traffic load/distribution variations, congestion, etc.).

F. Network scalability

Currently, popular network management algorithms were mostly designed to work on fixed or relatively small wireless networks. Many mobile ad hoc network applications involve large networks with tens of thousands of nodes, as found for example, in sensor networks and tactical networks [7]. Scalability is critical to the successful deployment of these networks. The steps toward a large network consisting of nodes with limited resources are not straightforward, and present many challenges that are still to be solved in areas such as: addressing, routing, location management, configuration management, interoperability, security, high capacity wireless technologies, etc.

G. Light-weight terminals

The nodes of ad hoc network are mobile devices with less CPU processing capability, small memory size, and low power storage. Such devices need optimized algorithms and mechanisms that implement the computing and communicating functions.

H. Energy constrained operation

Because batteries carried by each mobile node have limited power supply, processing power is limited, which in turn limits services and applications that can be supported by each node. This becomes a bigger issue in mobile ad hoc networks because, as each node is acting as both an end system and a router at the same time, additional energy is required to forward packets from other nodes.

IV. Power Saving Techniques

Since ad hoc networks do not assume the availability of a fixed infrastructure, it follows that individual nodes may have to rely on portable, limited power sources. The idea of energy-efficiency therefore becomes an important problem in ad hoc networks. Most existing solutions for saving energy in ad hoc networks revolve around the reduction of power used by the radio transceiver. At the MAC level and above, this is often done by selectively sending the receiver into a sleep mode, or by using a transmitter with variable output power (and proportionate input power draw) and selecting routes that require many short hops, instead of a few longer hops [4].

In mobile Ad hoc network there can be three aspects to reduce the power consumption.

- Power saving at mobile device level
- Power saving by controlling transmission level of packet
- Power saving by using optimized power routing protocol

A. Power saving at mobile device level

Mobile devices consume power even in their sleep mode For example, in mobile phones, even if they are not in use, there is a constant power drain because the trans-receiver is constantly hearing for signals to itself. A lot of efforts are currently going on to reduce the power consumed in each & every aspect of a mobile device. Now we give a brief description of some of these methods

- Disk scheduling

The operating system of a machine is responsible for using hardware efficiently — for the disk drives, this means having a fast access time and disk bandwidth. Access time has two major components seek time & Rotational latency. Seek time is the time for the disk are to move the heads to the cylinder containing the desired sector. Rotational latency is the additional time waiting for the disk to rotate the desired sector to the disk head. Disk bandwidth is the total number of bytes transferred, divided by the total time between the first request for service and the completion of the last transfer.

One method of energy conservation [8] in mobile devices is to spindown a disk in its idle time. The spindown delay is the amount of time the disk is idle before it spins down. [3] presents a quantitative analysis of the potential costs and benefits of spinning down a disk in its idle time. The tests were carried out using traces from both DOS machines and the Sprite File system. The conclusion was that the maximum power savings were obtained by using a spindown delay of two seconds as opposed to the 3-5 minutes recommended by most manufacturers. To justify this claim, the authors presented two points: frequency of sleep and length of sleep. They claim that, with shorter delays, the disk gets to sleep for a longer time and hence save more power.

The drawback of spinning down a disk after such short delays is the time and energy needed to spinup the disk, which results in user delay. Traces used by the authors show that the spindown occurs 8-15 times an hour. This translates to 16-30 seconds of user delay per hour, which is reasonable compared to the power savings incurred.

- CPU Scheduling

CPU scheduling is the basis of multiprogrammed operating systems. By switching the CPU among processes, the operating system can make the machine more productive. The power [8] consumed by a processor is directly proportional to the supply voltage, the switching capacitance of the various devices and the frequency of the clock. Gates in CMOS CPU's switch state at every clock cycle, which lead to a short circuit between the power-supply and ground. As a result more power is wasted with higher frequency.

The power required by the CPU is given by CV^2F, where C is the total capacitance of the wires, V is the supply voltage and F is the operating frequency. There are various algorithms proposed for adjusting the clock frequency in idle time. The main idea behind it is to balance the CPU usage between bursts of high utilization and idle times. Task or process scheduling can be an effective way of accomplishing this.

Almost all processes have a deadline by which they need to be executed. It has been observed in [6] that even when the processor is operating at the worst case, in scheduling the tasks, there is some idle time. This idle time is called the slack time. This slack time can be used to conserve energy by slowing down the processor and reducing the voltage. These techniques are known as, static slowdown and voltage scaling. We can reduce or eliminate the idle time by reducing the voltage to operate the processor such that, the process takes longer to finish but is completed before its deadline.

- Memory Allocation

Memory is the most important resource of a mobile device In mobile devices, memory instructions are among the highest consumers of power [3]. Since many small devices do not have a secondary storage, the power consumed by the memory is very crucial and needs to be optimized. Some of the memory devices like Direct Rambus DRAM (RDRAM), have come out with a DRAM that allows the individual devices to be in different power states. These devices are in decreasing order of power states and increasing order of access times: Active, Standby, Nap and Powerdown.

Memory Placement policies for code and data can also help to reduce the power consumption. If active pages with temporal locality are grouped together and placed on the same memory chip before moving to the next, the remaining chips can be powered down [3]. This technique helps in reducing the power consumed in reading data from memory. The simulation results given in [3] show power saving of about 6% - 50% using the static, dynamic and temporal locality placement policies.

B. Power saving by controlling transmit power level

The power control problem in wireless ad hoc networks is that of choosing the transmit power for each packet in a distributed fashion at each node. The problem is complex since the choice of the power level fundamentally affects many aspects of the operation of the network like:

1. The transmit power level determines the quality of the signal received at the receiver which affects the physical layer
2. It determines the range of a transmission which affects routing in terms affects network layer.
3. It determines the magnitude of the interference it creates for the other receivers which affects the transport layer due to congestion

Transmit power control is therefore a prototypical cross layer design problem affecting all layers of the protocol stack from physical to transport, and affecting several key performance measures, including the trinity of throughput, delay and energy consumption. Cross-layer design, in general, should be approached holistically with some caution, keeping in mind longer term architectural issues . Thus arises the question of where in the network architecture should power control be located, the resolution of which requires an appreciation of the issues involved at each layer.

- Design principles for power control protocol

Power control is important in wireless ad hoc networks for at least two reasons: It can impact on battery life, and It can impact on the traffic carrying capacity of the network.

Following are the design principles for power control.
1. To increase network capacity it is optimal to reduce the transmit power level.
2. Reducing the transmit power level reduces the average contention at the MAC layer.
3. The impact of power control on total energy consumption depends on the energy consumption pattern of the hardware.
4. When the traffic load in the network is high, a lower power level gives lower end-to-end delay, while under low load a higher power gives lower delay.
5. Power control can be regarded as a network layer problem.

So based on above design guidelines Kawadia & Kumar in [10] propose some protocols which attempt to achieve several design objectives and perform several optimizations simultaneously.

- The COMPOW protocol [10] attempts to increase network capacity, while meeting the needs of several other layers by choosing a common power level throughput the network.
- The CLUSTERPOW protocol [10] relaxes this constraint and provides a joint solution to the power control, clustering and routing problem, again with the goal of maximizing network capacity.
- The MINPOW protocol achieves a globally optimal energy consumption solution for awake nodes, but may or may not increase network capacity depending on the wireless hardware.

C. *Power saving by using optimized power aware routing protocol*

Routing is the process in which a route from a source to a destination node is identified and is achieved either by computing all routes before and presorting them or computing them when needed.

A routing protocol is a protocol that specifies how routers communicate with each other to disseminate information that allows them to select routes between any two nodes on a network. Typically, each router has a priori knowledge only of its immediate neighbors. A routing protocol shares this information so that routers have knowledge of the network topology at large.

In wireless ad hoc networks, every host acts both as a router and a packet sender, so the classical routing protocols used by wire linked networks are not applicable at all to ad hoc mobile networks. The routing protocols for ad hoc may be classified on the basis of following three criteria: Based on the logical organization, based on how to obtain routing information & based on how the routing path is created

- Based on the logical organization through which theprotocol "describes" the network

On the basis of the logical organization the routing protocols can be divided in "Uniform" and "Non Uniform" routing protocols.

In a uniform protocol, none of the nodes take on a distinguished role in the routing scheme: each sends and responds to routing control messages the same way. No hierarchical structure is imposed on the network. Although such a protocol avoids the resource costs involved in maintaining high-level structure, scalability may become an issue in larger networks.

Non-uniform protocol attempt to limit routing complexity by reducing the number of nodes participating in a route computation. Such an approach can improve scalability and reduce communication overhead; alternatively, it can support the use of algorithms of greater computational or communication complexity than is possible in the full ad hoc network. In addition, higher-level topology information can facilitate load balancing and QoS support.

- Based on the way routing information is obtained

From the routing information point of view, routing protocols may be divided in : Proactive (Table-Driven),Reactive (On-Demand) & Hybrid

- Proactive (Table-Driven)

In Table-driven routing protocols each node maintains one or more tables containing routing information to every other node in the network. All nodes update these tables so as to maintain a consistent and up-to-date view of the network. When the network topology changes the nodes propagate update messages throughout the network in order to maintain consistent and up-to-date routing information about the whole network. This type of protocols maintains fresh lists of destinations and their routes by periodically distributing routing tables throughout the network. Example of Proactive protocols are DSDV (Destination- Sequenced Distance-Vector), WRP (Wireless Routing Protocol) etc.

- Reactive (or On-Demand)

A different approach from table-driven routing is source-initiated on-demand routing. This is type of reactive routing creates routes only when desired the source node. When a node requires a route to a destination, it initiates a route discovery process within the network. This process is completed once a route is found or all possible route permutations have been examined. The route is perceived by a route maintenance procedure until either the destination becomes inaccessible along every path from the source or until the route is no longer desired.

In Reactive protocols a procedure is needed to establish the correct routing path only when packets are to be transmitted; in such a way signaling traffic is reduced, but with increasing delivery times.

Examples of Reactive protocols are AODV (ad hoc on-demand distance Vector), DSR (dynamic source routing) and TORA(temporally ordered routing algorithm)

- Hybrid

This type of protocols combines the advantages of proactive and of reactive routing. The routing is initially established with some proactively prospected routes and then serves the demand from additionally activated nodes through reactive flooding.
Examples of Reactive protocols are ZRP (Zone Routing Protocol)**,** HRPLS (Hybrid Routing Protocol for Large Scale Mobile Ad Hoc Networks with Mobile Backbones) etc.

- Based on how the routing path is created

Routing path is track the packet will follow from source to destination. From the routing path point of view the Protocols may divided into two categories: Source Routing & Non Source Routing.

In the first ones the sending node determines the complete path to the destination, registering it directly into the packet so, intermediate nodes only retransmit packets to those addressed by the directly into the packet so, intermediate nodes only retransmit packets to those addressed by the previously established path. In the latter, instead, the only routing information contained in data packets is that represented by the best neighbor node to which communication has to be forwarded; consequently, every node must be able to optimize routing decisions.

V. Conclusion

In this paper, we have given an overview of mobile ad hoc networks its features and investigated the problem of power saving in mobile ad hoc networks. We have studied current power saving techniques used at different levels .Power saving at routing protocols level is much easier as compared to, power saving at device level or transmission level. Each of these techniques saves some energy of mobile device and if we use these different techniques in a combined in a manner it saves lot of energy and increase the lifetime of network.

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Breeding Software Test Cases for Pairwise Testing Using GA

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***Abstract-*:Pairwise testing is a specification-based testing criterion, which requires that for each pair of input parameters of a system, each combination of valid values of these two parameters be covered by at least one test case [TAI02]. Pairwise testing has become an essential tool in a software tester's toolbox. This paper pays special attention to usability of the pairwise testing technique. In this paper, we propose a new test generation strategy for pairwise testing using Genetic Algorithm (GA). We compare the result with the random testing and find that applying GA for pairwise testing performs better result. Information on at least 20 tools that can generate pairwise test cases, have so far been published [PAI10]. Most tools, however, lack practical features necessary for them to be used in industry. Over the years, pairwise testing was shown to be an efficient and effective strategy of choosing tests [BUR98, BUR94, COH97, DUN97, KUH02, WAL01]. However, as shown by Smith et al. [SMI00] and later by Bach and Shroeder [BAC04] pairwise, like any technique, needs to be used appropriately and with caution.**

I. Introduction

Software testing is an important activity of the software development process. It is a critical element of software quality assurance. A set of possible inputs for any software system can be too large to be tested exhaustively. Techniques like equivalence class partitioning and boundary-value analysis help convert even a large number of test levels into a much smaller set with comparable defect-detection capability. If software under test can be influenced by a number of such aspects, exhaustive testing again becomes impracticable. A number of combinatorial schemes have been proposed to help testers choose subsets of input combinations that would maximize the possibility of finding errors. Random testing, Anti-random, Adaptive random testing and finally t-wise testing strategies with pairwise testing being the most famous among these (see Figure 1). Software system testing typically consists of only a very small sample from the set of possible inputs. It can be difficult or almost impracticable to generalize the test results from a limited amount of testing. It can also be very difficult to determine the nature and location of the errors. To address these issues, this paper presents genetic algorithm approach to breed software test cases for pairwise testing. In this article we investigate the performance of GA with different parameters combinations used to automate the test data generation. The work is compared with random testing and we concluded that the GA improves the search from one generation to the next, and performs better than random testing, where the search was absolutely random and does not show improvement through the generations. Another observation is that random testing generates less successful test cases than GA. Test data generation consists in proposing a superior set of input data for a program to be tested. This is a very significant, time consuming, and hard task in software development [KOR90]. But, what is a good set of data inputs? Intuitively, we can state that a good set of test data will allow a large amount of faults in a program to be discovered. For a more formal definition we have to resort to the test adequacy criteria [MIC01]. A great number of paradigms have been applied to the test data generation. A first paradigm is the so-called random test data generation. The test data are created randomly until the test adequacy criterion is satisfied or a maximum number of data sets is generated. Symbolic test data generation [CLA76] consists in using symbolic values for the variables instead of real values to get a symbolic execution. Some algebraic constraints are obtained from this symbolic execution and these constraints are used for finding test cases. Godzilla [OFF91] is an automatic test data generator that uses this technique. A third and widely spread paradigm is dynamic test data generation. In this case, the program is instrumented to pass information to the test generator. The test generator checks whether the test adequacy criterion is fulfilled or not. If the criterion is not fulfilled it prepares new test data to serve as input for the program. The test data generation process is translated into a function minimization problem, where the function is some kind of distance to an execution where the test criterion is fulfilled. This paradigm was presented in [MIL76] and there are many works based on it [KOR90, MIC01, JON96, WEG97]. Following

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the dynamic test data generation paradigm, several metaheuristic techniques have been applied to the [MAN05] present a recent review on the applications of the evolutionary algorithms to software testing. Most of the papers included in their discussion use GAs to find test data. In fact, only a few works listed in the review include other techniques such as cultural algorithms [OST99] (a special kind of GA), hill climbing [TRA00], and simulated annealing [TRA98]. We have found other recent works applying metaheuristic algorithms to software testing. In [DIA03] the authors explain how a Tabu Search algorithm can be applied to generate test data obtaining problem in the literature. Mantere and Alander in

maximum branch coverage. Sagarna and Lozano tackle the problem by using an Estimation of Distribution Algorithm (EDA) in [SAG03] and they compare a Scatter Search (SS) with EDAs in [SAG05]. In this work we propose the ES for finding input data sets in software testing. This technique has some advantages such as self-adaptability, and real number representation of the problem variables. The former reduces the human effort in tuning the algorithm. The last makes possible to explore a wide region of the input values.

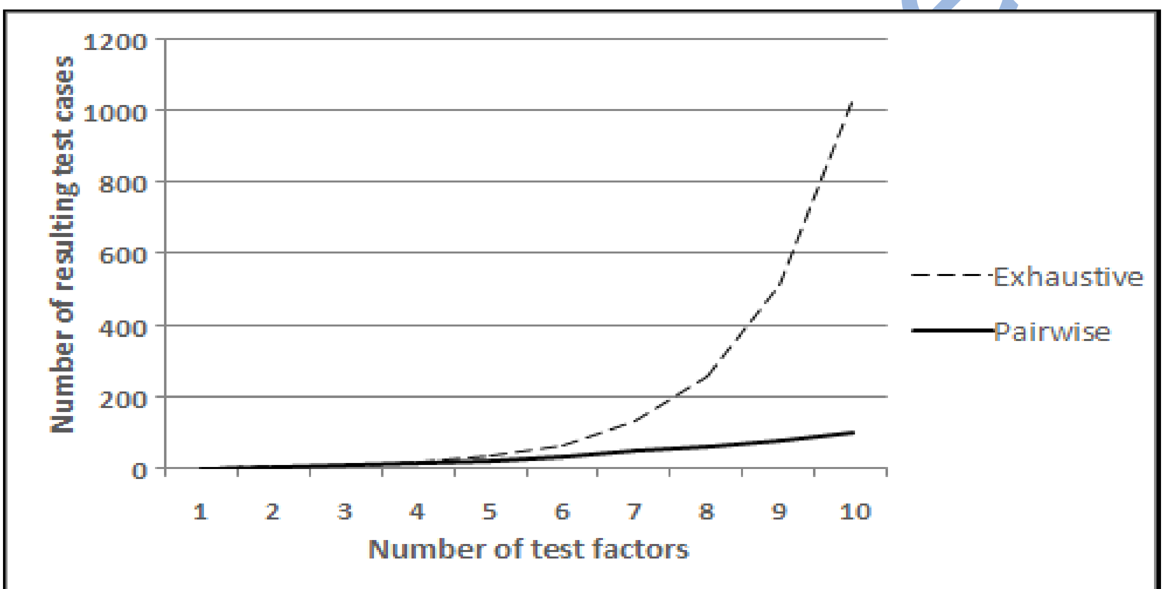

Figure 1: Increase in number of exhaustive and pairwise tests with number of test levels

II. Pairwise Testing

The necessary condition for pairwise testing is for each pairs of input parameters every combination of valid values of these two parameters be covered. Consider a system with parameters X, Y, Z and values (X1, X2), (Y1, Y2) and (Z1, Z2, Z3) respectively. For parameters X and Y the total number of possible pairwise test cases are limited to {(X1, Y1), (X1, Y2), (X2, Y1), (X2, Y2)} only but for parameters X, Y, Z a large number of pairwise test sets exists. Some test sets are as under.

1. {(X1, Y1, Z1), (X1, Y2, Z2), (X2, Y1, Z3), (X2, Y2, Z1), (X2, Y1, Z2), (X1, Y2, Z3)}
2. {(X1, Y1, Z1), (X1, Y2, Z1), (X2, Y1, Z2), (X2, Y2, Z3), (X2, Y1, Z1), (X1, Y2, Z2), (X1, Y1, Z3)}
3. {(X1, Y1, Z1), (X1, Y2, Z1), (X2, Y1, Z2), (X2, Y2, Z2), (X2, Y1, Z1), (X1, Y1, Z2), (X1, Y1, Z3), (X2, Y2, Z3)}

To illustrate the concept of pairwise testing Given a set of N independent test factors: f1, f2, f3 ...., fN, with each factor fi having Li possible levels: fi = {li,1, ..., li,Li}, a set of tests R is produced. Each test in R contains N test levels, one for each test factor fi, and collectively all tests in R cover all possible pairs of test factor levels i.e. for each pair of factor levels li,p and lj,q, where 1 <= p <= Li, 1 <= q <= Lj, and i <> j there exists at least one test in R that contains both li,p and lj,q. This idea can easily be extended from covering all pairs to covering any t-wise combinations where 1 <= t <= N. When t = 1, the strategy is equivalent to each-choice; if t = N (the maximum number), the resulting test suite is exhaustive.

III. Genetic Algorithm

Evolutionary testing is characterized by the use of metaheuristic search for test case generation. The considered test aim is transformed into an optimization problem [STH96] [STH01] where the input domain of the test object forms the search space for test data that fulfils the respected search aim. An example of evolutionary algorithms is Genetic Algorithms (GA). Genetic Algorithms, pioneered by John Holland, University of Michigan in 1970's and got popular in the late 1980's. The idea is purely based on Darwinian Evolution. It can be used to solve a variety of problems that not easy to solve using other techniques.

A genetic algorithm (GA) is a search algorithm based on principles from natural selection and genetic reproduction. GAs have been successfully applied to a wide range of applications, including optimization, scheduling, and design problems. Key features that distinguish GAs from other search methods include 1) A population of individuals where each individual represents a potential solution to the problem to be solved. 2) A fitness function which evaluates the utility of each individual as a solution. 3) A selection function which selects individuals for reproduction based on their fitness. 4) Idealized genetic operators which alter selected individuals to create new individuals for further testing. These operators, e.g. crossover and mutation, attempt to explore the search space without completely losing information that is already found. GA uses crossover and mutation operator to solve optimization problem using survival of the fittest idea. It is a search technique used in computing to find exact or approximate solutions to optimization and search problems. Genetic algorithms are categorized as global search heuristics. Genetic algorithms are a particular class of evolutionary algorithms (also known as evolutionary computation) that use techniques inspired by evolutionary biology such as inheritance, mutation, selection, and crossover (also called recombination). This technique can be applied to various problems, including those that are NP-hard. The technique does not ensure an optimal solution, however it usually give good approximations in a reasonable amount of time. This, therefore, would be a good algorithm to try to optimize the test cases. Genetic algorithms (GAs) are loosely based on natural evolution and use Darwin's principal of the "survival of the fittest" technique, where the best solutions survive and are varied until we get a good result. GA is based on the natural process of evolution. In nature, the fittest individuals are most likely to survive and mate; therefore the next generation should be fitter and healthier because they were bred from healthy parents. This same idea is applied to a problem by first 'guessing' solutions and then combining the fittest solutions to create a new generation of solution which should be better

than the previous generation. GA operates on a string of digits called chromosomes [BER03], each digit that makes up the chromosome is called gene, and a collection of such chromosomes makes up a population. Each has a fitness value associated with it, and this fitness value determines the probability of survival of an individual to the next generation. After the next generation is created a percentage of the chromosomes are crossed and small.

The random mutation elements are used to account for the occasional 'mishap' in nature. In GA there is a randomized exchange of structured information among a population of artificial chromosomes. When GAs are used to solve optimizations problems, good results are obtained surprisingly quickly. A problem is to maximize a function of the kind $f(x1, x2, \ldots xm)$ where $(x1, x2, \ldots, xm)$ are variables which have to be adjusted towards a global optimum. Three basic operators responsible for GA are (a) selection, (b) crossover & (c) mutation. The main genetic operator is crossover which performs recombination of different solutions to ensure that the genetic information of a child life is made up of the genes from each parent. The advantage of GAs is the fact that they are adaptive. Genetic Algorithms may be differentiated from more conventional techniques as (a) in GA a representation for the sample population must be derived, (b) GAs manipulates directly the encoded representation of variables, rather than manipulation of the variables themselves, (c) GAs use stochastic rather then deterministic operators, (d) GAs search blindly by sampling & ignoring all information except the outcome of the sample, (e)GAs search from a population of points rather than from a single point, thus reducing the probability of being stuck at a local optimum, which make them suitable for parallel processing. In the context of S/W testing, the basic idea is to search the domain for input variables which satisfy the goal of testing.

IV. Generating test suites using Genetic Algorithms

Using Genetic algorithms is one proposed way to test application [KAS96]. This method generates test cases based on the theory that good test coverage can be attained by simulating a novice user who would follow a more random path while an expert user of a system will follow a predictable path through an application ignoring many possible system states that would never be achieved. A novice user would follow an unexpected path to achieve the same goal so it's therefore more desirable to create test suites that simulate novice usage because they will test more. The difficulty lies in generating test suites that simulate 'novice' system usage. Novice paths through the system are not random paths. First, a novice user will learn over time and generally won't make the same mistakes repeatedly and secondly, a novice user is following a plan and probably has some domain or system knowledge. For the purposes of testing, each gene is essentially a list of random integer values of some fixed length. Each of these genes represents a path through the GUI. The success of the genes is scored by a criterion that rewards the best 'novice' behavior. In our approach the test cases are created incrementally using genetic approach, which synthesize query characteristics that are of interest for the purposes of test coverage. The algorithm selects one or more queries from the best query pool and uses any query mutation and combination techniques to create a new query.

Step1: Create initial population (randomly)
Step2: Evaluate individuals of population (fitness calculation)
Step3: Select individuals for mating(Crossover) to generate new population
Step4: Mutation
Step5: if stopping criteria do not satisfied goto step2
Step6: Stop

Figure1: Genetic Algorithm

Step1: Repeat step 1 to step 12
Step2:(Q1…..Qk)getQueriesFromBestQueryPool(randNum)
Step3: Q' generateNewQuery(Q1…..Qk)
Step4: resultSet (executeQuery(Q'))
Step5: if (resultSet = = empty) then
Step6: continue
Step7: end if
Step8: feedback (collectFeedback(Q'))
Step9: isFit = = true then
Step10: addReplaceQueryToBestQueryPool(Q'))
Step11: end if
Step12: Until timePeriod not expired

Figure2: Genetic Algorithm for Selecting Best Query from Query Pool

Two different function used in the above algorithm are Feedback and fitness. Feedback is used by the test case generator in the form of a set of strings describing the coverage achieved while executing meticulous query. On the other hand the fitness function decides whether query goes into best query pool or not. If query includes a gene that see for the first time, then that query always passes the fitness check and is added to the best query pool. Otherwise the fitness decision depends on whether a new query is shorter or faster than the existing query. Genetic Algorithms offer a 'one size fits all' solution to problem solving involving search. Unlike other conventional search alternatives, GA's can be applied to most problems out of the box, only needing a good function specification to optimize and a good choice of representation and interpretation. This, coupled with the exponentially increasing speed/cost ratio of computers, makes them a choice to consider for any search problem. **A** genetic algorithm is a heuristically guided random search technique that concurrently evaluates thousands of postulated solutions. Biased random selection and mixing of the evaluated searches is then carried out in order to progress towards better solutions. The coding and manipulation of search data is based upon the operation of genetic DNA and the selection process is derived from Darwin's ['Evolutionary Computation, D.Fogel1, IEEE Press 1995.] survival of the fittest'. Search data are usually coded as binary strings called chromosomes [Genetic Algorithms in Search, Optimization and Machine Learning, D. Goldberg, Addison-Wesley1989.], which collectively form populations. Evaluation is carried out over the whole population and involves the application of, often complex 'fitness' functions to the string of values (genes) within each chromosome. Typically, mixing involves recombining the data that are held in two chromosomes that are selected from the whole population. Genetic Algorithms have been intensively studied during the past three decades. Amounts of applications have benefited from the utilization of genetic algorithms. Theoretical approaches on genetic algorithms have also helped researchers to understand the mechanisms of genetic algorithms. This paper is produced in an attempt to provide **a** brief description of current developments in genetic algorithms.

The coverage percentage of pairwise testing using GA is as under.

Technique	Random	Pairwise	Pairwise GA
Coverage In TRN	60%	75%	85%
Coverage in OCT	65%	70%	80%

V. Conclusion

The shortcoming with the pairwise testing is that, if domains of input parameters are huge the number of generated test sets became extremely large. For a system with each parameter having N values, the number of tests required for pairwise testing is at least N^2. Consequently, if each parameter has 1000 values, at least one million tests are required for implementing pairwise testing technique. To alleviate this test suites explosion problem we can split each input domain into partitions, choose one agent value from each partition, and breed tests according to agent values for input parameters. By controlling the number of partitions for each input parameter, the number of tests needed for pairwise testing can be determined.

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On An Alternative Method Of Estimation Of Polynomial Regression Models

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Abstract- **The OLS and ML methods had been important tools of obtaining estimates of parameters in regression analysis. This paper proposed a numerical analytic method as an alternative approach to the OLS method for a Polynomial Regression Model of the third degree.The model was fitted to a data of output and total production cost of a commodity using the OLS and the computational group average (of numerical analysis) techniques.We obtained very close parameter estimates in both cases.**

KEYWORDS: OLS-Ordinary Least Square, ML-maximumlikelihood,Regression,Parameters, Computational Group Average

I. Introduction

Statistical models have numerous real life applications in all professional walks of life. One of them is found in econometrics meaning econometric measurements. The models according to functional forms are categorized into linear and non-linear models. The polynomial regression models are important class of non-linear models which have extensive use in econometric researches especially situations relating to cost and production. Its non-linearity came from the fact that the relationship between marginal cost (MC) of production(Y) of a commodity and its output (X) is non-linear since a U-shaped curve (parabola) is usually observed in its scattered diagram. This of course, represents a quadratic form. That is, a polynomial regression model of degree two whose quadratic form is given by

$$Y_i = \beta_0 + \beta_1 X_i + \beta_2 X_i^2 + \varepsilon_i \quad , \quad i = 1, \ldots, n \qquad (1)$$

X_i is assumed to be fixed called the independent variable and consequently X_i^2 is also fixed while ε_i is the random error term. β_0 , β_1 , and β_2 are the coefficients of the equation.

Its general form is

$$Y_i = \beta_0 + \beta_1 X_i + \beta_2 X_i^2 + \cdots + \beta_k X_i^k + \varepsilon_i \qquad (2)$$

called the polynomial regression model of degree k.

The exponential, logistic and Gompertz models are also important class of non-linear regression models with great applications in growth and population studies having respective forms;

$$Y_i = \beta_1 e^{\beta_2 X_i} + \varepsilon_i \qquad (3)$$

$$Y_i = \frac{\beta_1}{1 + \beta_2 e^{-\beta_3 X_i}} + \varepsilon_i \qquad (4)$$

$$Y_i = \beta_1 e^{-\beta_2 X_i} + \varepsilon_i$$

Often times, X_i is replaced by t_i in (3)-(5) since Y_i represents growth in relation to time t_i.

II. Materials and Methods

A data of output and total cost of production of a commodity in the short run is considered below

Output	1	2	3	4	5	6	7	8	9	10
TPC	193	226	240	244	257	260	274	297	350	420

The cattered diagram is given by the succeeding graph

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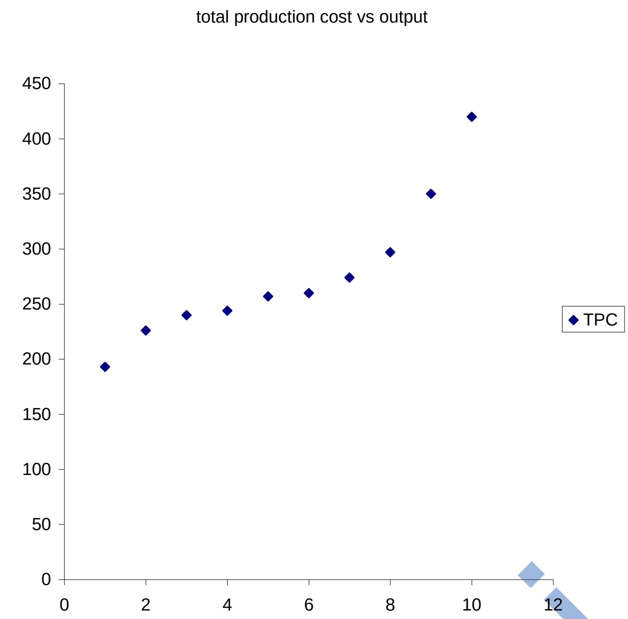

Scattered diagram of output and total production cost

We observed an S-shaped curve which follows the principle of the celebrated law of diminishing returns in the short run. This is identical to a cubic function.

We therefore propose a polynomial regression model of the third degree for the data, so that

$$Y_i = \beta_0 + \beta_1 X_i + \beta_2 X_i^2 + \beta_3 X_i^3 + \varepsilon_i \qquad (6)$$

III. THE OLS METHOD

The ordinary least square method of estimation in regression analysis is used as follows since the model does not violate the non multicollinearity assumption.

Then,

$$\sum_{i=1}^{n} \varepsilon_i^2 = \sum_{i=1}^{n} \left(Y_i - \beta_0 - \beta_1 X_i - \beta_2 X_i^2 - \beta_3 X_i^3\right)^2 \qquad (7)$$

taking the partial derivatives of equation (7) w.r.t. the parameters β_0 , β_1 , β_2 , and β_3 and setting the resulting results to zero. These yield;

$$\frac{\partial \sum \varepsilon_i^2}{\partial \beta_0} = (-2)\sum\left(Y_i - \beta_0 - \beta_1 X_i - \beta_2 X_i^2 - \beta_3 X_i^3\right) = 0 \qquad (8)$$

$$\frac{\partial \sum \varepsilon_i^2}{\partial \beta_1} = (-2X_i)\sum\left(Y_i - \beta_0 - \beta_1 X_i - \beta_2 X_i^2 - \beta_3 X_i^3\right) = 0 \qquad (9)$$

$$\frac{\partial \sum \varepsilon_i^2}{\partial \beta_2} = (-2X_i^2)\sum\left(Y_i - \beta_0 - \beta_1 X_i - \beta_2 X_i^2 - \beta_3 X_i^3\right) = 0 \qquad (10)$$

$$\frac{\partial \sum \varepsilon_i^2}{\partial \beta_3} = (-2X_i^3)\sum\left(Y_i - \beta_0 - \beta_1 X_i - \beta_2 X_i^2 - \beta_3 X_i^3\right) = 0 \qquad (11)$$

We note that equations (8) - (11) can also be written as

$$\sum \varepsilon_i = 0 \qquad (8a)$$

$$\sum X_i\ \varepsilon_i = 0 \qquad (9a)$$
$$\sum X_i^2\ \varepsilon_i = 0 \qquad (10a)$$
$$\sum X_i^3\ \varepsilon_i = 0 \qquad (11a)$$

Equations (8a) – (11a) show that the properties of least squares fit namely that the residual is equal to zero and they are uncorrelated with the independent variables.

We solve (8) - (11) further to obtain their normal equations given by equations (8b) – (11b) so that;

$$\sum Y_i = n\beta_0 + \beta_1 \sum X_i + \beta_2 \sum X_i^2 + \beta_3 \sum X_i^3 \qquad (8b)$$
$$\sum X_i\ Y_i = \beta_0 \sum X_i + \beta_1 \sum X_i^2 + \beta_2 \sum X_i^3 + \beta_3 \sum X_i^4 \qquad (9b)$$
$$\sum X_i^2\ Y_i = \beta_0 \sum X_i^2 + \beta_1 \sum X_i^3 + \beta_2 \sum X_i^4 + \beta_3 \sum X_i^5 \qquad (10b)$$
$$\sum X_i^3\ Y_i = \beta_0 \sum X_i^3 + \beta_1 \sum X_i^4 + \beta_2 \sum X_i^5 + \beta_3 \sum X_i^6 \qquad (11b)$$

Fitting the data of output and total production cost to equations (8b) – (11b) and solving for the unknown parameters, we obtain the OLS estimates

$$\hat{\beta}_0 = 141.7667\,,\ \hat{\beta}_1 = 63.4776,\ \ \hat{\beta}_2 = -12.9615\,,\ and\ \hat{\beta}_3 = 0.9396\,.$$

Hence, we have

$$Y = 141.7667 + 63.4776\mathrm{X} - 12.9615\mathrm{X}^2 + \ 0.9396\mathrm{X}^3 \qquad (12)$$

called the fitted curve.

<u>The Numerical Approach</u>

We begin by writing (6) as

$$Y = \mathrm{F(X)} = \beta_0 + \beta_1\mathrm{X} + \beta_2\mathrm{X}^2 + \ \beta_3\mathrm{X}^3 \qquad (13)$$

Proceeding with the application of the Computational Group Average scheme, we assume that a particular point $(X_1\,, Y_1)$ satisfies the curve represented by (13).

That is,

$$\mathrm{Y}_1 = \beta_0 + \beta_1\mathrm{X}_1 + \beta_2\mathrm{X}_1^2 + \ \beta_3\mathrm{X}_1^3 \qquad (14)$$

Subtracting (14) from (13), we have

$$Y - Y_1 = \beta_1\left(X - X_1\right) + \beta_2\left(X^2 - X_2^2\right) + \beta_3\left(X^3 - X_3^3\right) \qquad (15)$$

Dividing (15) through by $x - x_1$, this yields

$$\frac{Y - Y_1}{X - X_1} = \beta_1 + \beta_2\left(X + X_1\right) + \beta_3\left(X^2 + XX_1 + X_1^2\right) \qquad (16)$$

We set $\dfrac{Y - Y_1}{X - X_1} = y$, $X + X_1 = x$ and $\left(X^2 + XX_1 + X_1^2\right) = z$ into (16) so that

$$y = \beta_1 + \beta_2 x + \beta_3 z \qquad (17)$$

Taking the first point (1,193) from the data as the particular point that satisfies the curve and substituting into (16), then

$$\frac{Y - 193}{X - 1} = \beta_1 + \beta_2\left(X + 1\right) + \beta_3\left(X^2 + X + 1\right) \qquad (18)$$

We divide the data into three groups since we are seeking the estimates of three unknowns β_1 , β_2 *and* β_3 for now. The groups are namely:

Group I

X	Y	z	x	y
1	193	-	-	-
2	226	7	3	33
3	240	13	4	23.5
4	244	21	5	17
	Totals	41	12	73.5
	Means	13.6667	4	73.5

Group 2

X	Y	z	x	y
5	257	31	6	16
6	260	43	7	13.4
7	274	57	8	13.5
	Totals	131	21	42.9
	Means	43.6667	7	14.3

Group 3

X	Y	z	x	y
8	297	73	9	14.86
9	350	91	10	19.63
10	420	111	11	25.22
	Totals	275	30	59.71
	Means	91.6667	10	19.9033

Putting all the means for the three groups into (17), these yield

$$24.5 = \beta_1 + 4\beta_2 + 13.6667\beta_3$$

(19)

$$14.3 = \beta_1 + 7\beta_2 + 43.6667\beta_3$$

(20)And;

$$19.9033 = \beta_1 + 10\beta_2 + 91.6667\beta_3$$

(21)

which constitutes a system of linear equations with three unknowns.

By putting (19) – (21) in matrix form and solving for the unknowns, using Gaussian Elimination method, we obtain

$$\beta_1 = 62.2206 \,,\ \beta_2 = -12.18 \,,\ \text{and}\ \beta_3 = 0.8780$$

We solve for β_0 by putting the values of $\beta_1 \,, \beta_2 \text{ and } \beta_3$ into (18), so that

$$\frac{Y-193}{X-1} = 62.2206 \text{ -}12.18\left(X+1\right) + 0.8780\left(X^2 + X + 1\right)$$

(22)

On expanding (22) and solving for Y, then

$$Y = 0.8780X^3 \text{ -}12.18X^2 + 62.2206X + 142.096$$

That is,

$$Y = 142.096 + 62.2206X - 12.18X^2 + 0.8780X^3$$

which is identical to the estimated production cost function curve in (12)

IV. Results and Discussions

The OLS approach gave parameter estimates $\hat{\beta}_0 = 141.7667 \,,\ \hat{\beta}_1 = 63.4776,$ $\hat{\beta}_2 = -12.9615 \,,\ \textit{and}\ \hat{\beta}_3 = 0.9396$ giving rise to an estimated curve $Y = 141.7667 + 63.4776X - 12.9615X^2 + 0.9396X^3$.

The numerical approach gave unknown parameter values $\beta_0 = 142.096 \,,$ $\beta_1 = 62.2206 \,,\ \beta_2 = -12.18 \,,\ \text{and}\ \beta_3 = 0.8780$ which also led to the curve $Y = 142.096 + 62.2206X - 12.18X^2 + 0.8780X^3$.

By comparison, we observed that the values of the parameter estimates are very close to one another in both cases but however not exactly the same. This could be due to the errors incurred by the varying assumptions in the underlying principles of both methods.

The results *obtained* by the numerical approach also satisfy the curve. Hence, the numerical approach is equally accurate and consequently valid.

V. Conclusion and Recommendation

We conclude that the computational group average (numerical) approach is valid and we recommend it as an alternative approach for estimation of polynomial regression models. We also recommend it for handling estimation in other non-linear models especially cases of small samples.

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Risk Mitigation And Management Scheme Based On Risk Priority

Basit Shahzad[1]Sara Afzal Safvi[2]

Abstract **-Much effort has been put in order to identify the possible risks hindering the successful completion of software projects. Techniques in risk mitigation, management and monitoring plan devise the estimation process of risk likelihood and their possible impact on the progress of software project. Risk Mitigation, Monitoring, Management is a thorough and continuous process, which aims to bring the potentially losing project to the safer shore. Hence every phase of this plan is of equal importance. Generally more focus is maintained in the initial phases i.e. the identification and assessment of possible risks. Whereas formalizing a concrete avoidance / mitigation plan must also be devised, to ensure that risk do not mature in problem. A response should be ready in advance. Generally it is easier to indentify and assess the risk but to suggest suitable mitigation / contingency plan is far more difficult task.**

The measurement of effectiveness of these mitigation / contingency plans should be well carried. It must ensure that after the execution of such plans the risk exposure is reduced or preferably eliminated. This can be referred as the feasibility of the mitigation / contingency plan, which is critically analyzed and measured for its effectiveness. This paper focuses on the prioritization and then handling and proposing the mitigation strategy for each risk factor

Keywords-Risk mitigation, Risk Priority, Risk Management, Risk Handling, Software Risk management

I. Introduction and Risk Categorization

Risk can either be avoidable or unavoidable. Hence risks can be categorized into two main classes. Based upon the priority level of any risk we can also judge if this risk is avoidable or not. So we can either build up mitigation or avoidance plan for that specific risk. [Table 1 shows the most prominent risks which may hinder the successful project completion].

Any Risk may be same in type but it may differ in different kind of software projects. For example Low estimation of cost may result differently in an embedded system software project where as it may behave differently in an Information System software project. It would be more convenient to calculate the cost of an embedded system. The reason is that embedded system would be having limited environment of functionality and narrow area of integration. Whereas as compared to this an information system would be involving lot of user types, broader integration area and vast environment (e.g. distributed enterprise systems)

Risk	Probability	Part-Impact A	D	C	T	M	Avg-Impact	Effected factors	Impact	Impact's probability	Overall Impact
1. Requirements are not properly stated	50%	1	2	3	4	5	3	2,3,7	45	22.5	55.2
2. Low estimation of cost	50%	3	3	3	3	3	3	8,9,10, 11,12, 17	45	22.5	103.9
3. More stress of users than expected	30%	0	0	0	2	4	2		12	3.6	3.6
4. Less reuse than expected	30%	0	0	3	0	0	1	2,17	3	0.9	57.4
5. Delivery deadline tightened	30%	4	4	5	4	4	4	7,10,11,15	84	28	136
6. Funding will be lost	10%	3	3	3	3	3	3	10,11	45	45	41.7
7. Technology does not meet expectations	30%	1	1	5	3	1	2		22	6.6	6.6
8. Lack of training on tools	10%	2	2	2	2	2	2	4,9,12	20	2	11.2
9. Staff inexperience	10%	2	2	2	3	2	2	1,2,4,6,7,8,13, 15,17, 18	22	2.2	145.6
10. Staff turnover	30%	3	3	5	4	4	4	6, 17	76	22.8	61.3
11. Manager changes circumstances	40%	3	3	2	2	2	3	2,17,18	36	14.4	92.5
12. Backup not taken	20%	2	2	4	1	1	3	13	30	6	15
13. Actual data/document loss	20%	3	3	3	3	3	3	2,16,17	45	9	69.7
14. Flood, fire and building losses	10%	2	2	2	2	2	2	2,17	20	2	58.5
15. Too many development errors	50%	0	0	0	5	5	4	6,10,11,17	40	20	95.7
16. Developer run away with code/doc	10%	0	0	5	4	4	3	2,6,17	42	4.2	65.2
17. Low estimation of time	50%	4	4	4	5	4	4	1,2,6	68	34	83.5
18. Lack of intuition	30%	4	4	3	4	3	4	1,2,4, 15	72	21.6	69.5

Table 1 Risk probability and over all impact [7]

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Therefore proposed mitigation and management plan would differ based upon project type. In this paper either mitigation or management plan is proposed for each type of risk based upon the fact of avoidance and un-avoidance.

A. Requirements are not properly stated

Yet a major risk factor affecting project schedule, budget and quality is the ability to successfully elicit requirements and execute on them.
This risk is avoidable and can be mitigated right from the beginning if tight grip is maintained in requirement elicitation phase.

II. Mitigation Strategy

1. Maintain clear understanding of stakeholder needs and their relative prioritization.
2. Bridge the communication gap between customer so they can not claim about their requests being misunderstood resulting in rework of demands to be implemented.
3. Divide the users into specific groups based upon organizational hierarchy and target each group separately. As each group would be having different requirement based upon their set of duties. For example managers would be interested in decision support tools where as operational work force would be interested in convenient transaction processing.
4. Use every possible method to understand what user says and what the analyst comprehends out of that, so that the requirement shall be validated by the end user.
5. Prototype Demos and screen shot can be showed to the user to avoid ambiguities. Confirmed user requirements shall be document and signed by all stake holders.
6. All stake holders specially customer must be told very clearly about the feasibility of particular requirement.
7. Joint Application Development (JAD) is a group based requirement elicitation and design technique. JAD mainly features an intensive structured workshop. Expected end users, Analysts, Developers and Projects managers attend the workshop. The workshop is headed by an experienced leader. The leader conducts meetings with managers and end users to clearly define the domain, scope and objectives of the project. This leader also determines participants of a JAD workshop. The output of this workshop is a document which contains the clear user interests determined during JAD session. [12]
8. Facilitated Application Specification Techniques (FAST) aim to further decrease ambiguities in requirement elicitation process. As developers and customer work as team member rather than behaving as opponent parties. This attitude generally ends up in confrontation and confusions which hinder the clear elicitation of user requirements. FAST brings a facilitator between the customer and developer who conducts a meeting and behaves as a mediator.

"During FAST meetings the following activities take place.

a) Product need and justification
b) Lists discussed and combined
c) Lists refined
d) Mini-specifications prepared for each list entry
e) Mini-specifications reviewed by all
f) Validation criteria for the product/system" [10]

A. Low estimation of cost

Accurate cost estimation is still a bottleneck in software planning process. Several methods already exist for this purpose. Mainly there are two categories of models to estimate the cost of software projects i.e. Algorithmic and non-algorithmic. Moreover most of the models are based upon the size of software project to calculate the cost. Each model has its own strengths and weakness. Selection of model revolves around the accuracy of its estimates. Unfortunately the accuracy of these models is not satisfactory. Moreover accurate cost estimation is the biggest success factor as well as risk in software development cycle. Software cost estimation focuses upon three main dimensions i.e. Human Effort, Time Duration and monetary resources required.
Keeping the unsatisfactory level of accuracy of existing cost estimating models, one must carefully decide which software cost estimation model to use. Which software size measurement to use (lines of code (LOC), function points (FP), or feature point). A good estimate must be determined keeping the project characteristics in view.

B. Mitigation Strategy

I. This risk can be mitigated by early selection of best available cost estimation model relevant to the project characteristics. *Empirical Cost Estimation model* utilizes the historical data about past projects. Therefore it shall be evaluated to confirm if sufficient historical data about the same kind of projects (similar processes, similar technologies, similar environments, similar people and similar requirements) is available or not.
II. Expert judgment shall not be relied too much upon, as poor measurement of project size may result in unrealistic cost estimation.
III. *Analytical model* is another alternative, which is based upon the rate at which developer solve problems and the number of problems available. Line Of Code or Function Points are analyzed for project size measurement. Realistically it is very

hard to calculate the actual number of code lines prior to the completion of project. Therefore if inaccurate size is input then, obviously inaccurate cost estimate would be resulted.[2]

IV. The relationship between cost and system size is not linear. Cost tends to increase exponentially with size. The expert judgment method is appropriate only when the sizes of the current project and past projects are similar.

V. Size of the project shall be measured keeping the detailed Work Breakdown Structure. So that cost estimate may encompass every area of cost and effort.

VI. Whichever cost estimation model is selected, it must be approved by all the stakeholders. Moreover the inaccurate factor shall be clearly explained to all stakeholders so that in future any drift from the planned estimated can be justified without contention.

VII. Experienced cost estimators shall be appointed to avoid any errors during process of cost estimation by inexperienced estimators.

VIII. Accurate measurement of project size is very essential as it leads to the accurate cost estimation of the project. [2]

C. *Contingency Plan*

Once the risk has matured into problem contingency plan can be executed for recovery.
Best approach can be to execute another iteration of cost estimation for overrun project. As proposed in [3].

I. If planned duration is overrun, then compression techniques can be followed by squeezing the activities on to the Critical path. [11]

II. The manager must analyze the reason of delay, if it might have been caused by inefficiency of the workers. The penalty shall be placed on them in terms of over time with out extra payment, therefore saving any further monetary cost overrun.

III. More stress of users than expected

This type of risk is surely avoidable and has the least overall impact ratio as depicted in table 1.
User sometimes may behave differently and hence may produce difficulties. This may result in more pressure on the development of the project.

A. *Mitigation Strategy*

I. If ample level of understanding has been developed with the user, then user must not stress upon unrealistic demands. Once requirements are clearly documented, the user may not find any capacity to further argue about the requirements unnecessarily.

II. Non functional Prototype can be shown to the user, if user becomes hyper to see the progress of the project.

IV. Less reuse than expected

Initially the cost of project might have been calculated by considering a good ratio of software reuse. And eventually it may not be practiced in reality which emerges into a severe risk factor.
This is an avoidable risk therefore can be planned to mitigate as earliest as possible.

A. *Mitigation Strategy*

I. The source of reusable software shall be determined before actually using the software. Following factors must be carefully analyzed first to see :
   a) If re useable software is available in house.
   b) If any contract is made with third party to provide re useable software.
   c) If any cost is incurred for re using the software, shall be carefully estimated.

II. Once the source is well specified, it can be assured that software / re useable components would be well in time available. Specially when they are not available in house.

III. Any delayed provision of such components shall be compensated by the third party, which is in contract.

V. Delivery deadline tightened

This risk has the second highest risk impact (i.e. 136) as depicted in table 1.
This may mature in delayed project delivery, therefore must be handled in time. Hence it may be avoidable, but once mature the contingency plan is proposed.

A. *Mitigation Strategy*

I. Close project monitoring shall be implemented continuously throughout each phase of the project.

II. Even if project is over running a single day, workers shall work over time to recover in time.

III. Project manager must make sure that software process is followed strictly.

IV. Moreover the entire organization must have matured to senior levels of CMMI or ISO whichever quality assurance process is implemented.

V. It is evident that organizations who have well achieved maturity level can better avoid risk at the initial stages.

VI. Proper tools and methods of configuration management shall be well in practice so that any requirement change may well incorporated and may not result in the delay or schedule tightening.

B. Contingency Plan

Tightened schedule would definitely result into pressure. As gone time can not be reversed. Therefore time loss can not be recovered rather extra burden falls on to the shoulders of the workers.
Although an iteration to revise the schedule can be made to increase the number of working hours per day and completing the work in restricted time slot. [3].

VI. Funding will be lost

Before taking off the fuel tank must be assured for fullness. Non availability of the funding can result in catastrophic results, similar to the crash of flight. Therefore ample funding should be guaranteed.
Although this risk has 10% probability factor but if turns into reality then it may earn total bad name and irrecoverable project failures.

A. Mitigation Strategy

I. The sources of project funding must be determined and agreed upon by all the stake holders in the very initial stages of the project life cycle. Rather at the feasibility study stage.
II. If project is financed by some bank loan then all the necessary terms and conditions should be in place and well documents.
III. It is better to have some insurance plan for contingency effort.

VII. Technology Does Not Meet Expectations

This may effect the re usability factor as well. Some reusable components, which best fit the user requirement, are not adapted for technological incompatibility issues.
There can be lots of issues in this regard:

I. Insufficient skilled human resource for that specific technology.
II. More funding is required.
III. Difficult maintainability.
IV. Incompatibility with other components.
V. Evolution is not possible.
VI. Customer resistance for the technology.

D. Mitigation Strategy

I. Selection of technology is done at the very beginning therefore all above mentioned issues must be addressed.
II. A checklist should be made and a thorough comparison should be carried out to determine the best suited technology.
III. All stake holders must be taken in confidence for the use of specific technology.

VIII. Lack of training on tools

Lack of training can be compensated by different strategies which may avoid this risk. This is minor risk as depicted at the second lowest number in risk priority table. But surely may not be underestimated to carry its impact to next stages, which may eventually result in delay due to in experienced workers.

A. Mitigation Strategy

I. If funding is low for the project, then project manager may compromise over less experienced staff, but it must be supplemented by in house staff training prior to the work starts.
II. Activity slacks can be utilized for the training of next task.

IX. Staff Inexperience

This may also prove deadly for the project success. As at any stage delay can be caused by mishandling of tasks by inexperienced workers.

A. Mitigation Strategy

I. Team members selection shall be done very carefully selecting only those workers who have good experience on the tools.
II. Experienced staff should be allocated to critical task which may ensure that no delay is expected and hence ensure the smooth and efficient completion of the project.

X. Staff turnover

This risk may be rooted very deep in the psychology of the workers. Many of the factors including internal and external to the organization can affect the throughput of the workers. Though external factors can not be fully controlled but at least internal factors can be eliminated or either minimized.

A. Mitigation Strategy

I. Workers should have strong motivation for work in terms of monetary or other rewards.
II. Workers should be appreciated for what ever effort they put in the project.
III. In case workers deliberately ignore their responsibilities, some kind of penalty shall be placed on them. Therefore there shall be some balancing threat to make them work.
IV. Close monitoring shall be kept to know the status of work done by each individual and obtain any kind of possible hurdles (e.g. sick leave, resignation plan).

XI. Manager changes circumstances

Rescheduling may bring many disturbance for the workers as their personal life may also be effected. An other cause may be forcefully switch over to different tool may upset the worker and may shatter the confidence level. All these factors can be avoided.

A. Mitigation Strategy

I. A software house should well maintain its team of skilled workers (i.e. Analysts / Programmers / testers).
II. Different programmer teams can be built for specific tools. Therefore a foot ball player shall not be forced to play tennis.
III. Managers must adapt allocation of tasks to such individual who are extremely confidant to carry out that task.
IV. A substitute worker shall always be spared so that in case of extra burden of over time can evenly be distributed among them.
V. Incase there are many work places scattered over the globe or nation wide, worker should be sent to the place of his/her desire. Parting from family or social circle may also disturb the worker emotionally and mentally.
VI. If project manager feels any discomfort in any of the worker, he/she must adapt an empathetic attitude towards him/her. And must try to find out the real root cause and may try to resolve the problem if possible.

XII. Backup Not Taken

In an information and technology based organization, it is next to a folly not to take regular back ups for the precious data resources.
Back up is not only necessary for data recovery, but Project management and configuration management also rely on the data about all stages.

A. Mitigation Strategy

I. Back up should be taken on regular basis.
II. Some authority shall confirm that back ups are taken regularly and intermediate versions of data are not ignored or lost.
III. More over back ups can be kept at multiple places. For this multiple back up servers can be employed at different geographical locations.

XIII. Actual Data/Document Loss

Although it is an irrecoverable loss, yet can surely be avoided.

A. Mitigation Strategy

I. Back up shall be kept not only in the office building but at some other place as well so that in case of any natural disaster, it shall be recovered.
II. Data and software library are assets of the organization therefore shall be valued and accordingly legal documentations shall be maintained in case are stolen or deliberately damaged.

XIV. Flood, Fire and Building Losses

Natural disasters can not either be avoided nor informed before. Therefore any lost caused by such threats must be born and there shall some concrete contingency planning for them.

E. Contingency Plan

I. Company assets must be insured to retrieve the loss.

XV. Too Many Development Errors

Development errors are natural to occur, but frequency should not exceed from a reasonable rate. This risk may be avoided successfully but detection of errors is not an easy task until the software is put through the testing phase.
Therefore this may come to the surface at the later stage i.e. testing phase.
Hence a mitigation as well as contingency plan can be devised for this risk.

A. Mitigation Strategy

I. Employment of experienced programmers can prevent too many errors in the code modules.
II. Moreover an experienced programmer can produce a better piece of code, which can be more efficient in logic implementation and reusability.
III. Lots of errors can be detected earlier, therefore code must be tested concurrently for such types of errors.

B. Contingency Plan

I. Errors can result in worst loss if detected after deployment of the system at the user end, as cost to fix errors after deployment is too high. Hence all possible errors should be tested and verified carefully prior to system delivery.
II. If error ratio is too large, the coders / testers may be put to over time to recover the errors.

XVI. Developer Run Away With Code/Documents

This can surely be avoidable risk. And can be prevented following the below mentioned measures:

A. Mitigation Strategy

I. Whenever some new employee is hired, a contract shall be signed clarifying the ownership of the code / design created by the employee.
II. There shall be some surety bond filled by the employee that he/she may not take away the technical material or shall not sell to other outside parties.

III. If so, there shall be some legal penalty to prevent such theft.
IV. Good configuration management shall be in place.

B. Contingency Plan

I. Proper configuration management should be practiced so that if latest version is lost then at least one previous version remains available. So that project can be resumed from one step behind.
II. This may cause in little tightening of the schedule and therefore shall be prevented by the rescheduling of the work by putting over time effort.

XVII. Low Estimation of Time

Likelihood of this risk is high as much as 50%, therefore the impact can effect the successful completion reasonably. Time estimation is as much complicated factor as cost estimation and faces many of the inaccuracies.
Hence the same sort of precautions shall be adapted as mentioned in Low Cost estimation risk section.

XVIII. Lack of intuition

This factor may vary on individual basis. As some veteran project manager would be able to sense the likelihood of problem occurrence without any evidence yet emerged onto the surface.

A. Mitigation Strategy

I. Inexperienced project manager should not be granted the steering of the ship. As captain of the ship must be strong nerved and must have a foresight to cope up with any problems hindering the smooth sailing.

XIX. Conclusion

In this paper major focus is put to devise and suggest an effective response towards a risk so that it can be prevented rather then the need of cure. Much work has been done to asses the risks, but few relates to the development of accurate responses to the risks.
We have also investigated mitigation and contingency strategies considering the priority level of each risks and the likely frequency and effect of each risk in any or all of the phases of software development life cycle.
This work is a contribution towards risk avoidance and a remedy measure is proposed for each type of risk. It has been observed that a large ratio of the software projects fail due to many risk factors. Those risk factors have been clearly identified and assessed many times. But still these risks mature into problems causing the project failure. We have proposed solutions and mitigation plans against each type of risk focusing at its specific priority in the risk listing and the probability. Mitigation and contingency plan may reasonably be affected by the likelihood and impact factor of each risk. Based upon these the risks have been prioritized [7]. Risks with smaller likelihood but greater impact or vice versa are equally important to be mitigated and controlled. Therefore risk priority can determine the importance of any mitigation or contingency plan to be activated. The timing of activation can also be determined by examining the risk priority.
This is essential to know that how many phases of the software development life cycle those plans should be spanning over. Risk may not be easily got rid off, it may decrease its likelihood in one phase and may eventually catch momentum in the other.
This paper may serve for the basis to further improve the risk mitigation and management strategies.

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Analyzing Zone Routing Protocol in MANET Applying Authentic Parameter

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Abstract- **A Mobile Ad-Hoc Network (MANET) is a decentralized network of autonomous mobile nodes able to communicate with each other over wireless links. Routing is the main part of wireless ad-hoc network conventionally there are two approaches first one is Proactive and another one is Reactive. The Zone Routing Protocol (ZRP) is a hybrid routing protocol for MANET, which able to proactively maintains routes within a local region of the network called as routing zone. Knowledge of this routing zone topology is leveraged by the ZRP, to improve the efficiency of a reactive route query/reply mechanism. The ZRP can be configured for a particular network through adjustment of a single parameter, the routing zone radius. ZRP uses the proactive and the reactive routing according to the need of the application at that particular instance of time depending upon the prevailing scenario. This work revolves around the performance of ZRP against realistic parameters by varying various attributes such as Zone Radius of ZRP in different node density. Results vary as we change the node density on Qualnet 4.0 network simulator.**

Keywords: MANET, ZRP, Routing Protocol, Routing Zone, proactive routing, reactive routing, hybrid routing

I. Introduction

Mobile ad hoc networks (MANETs) [1] are collections of mobile nodes, dynamically forming a temporary network without pre-existing network infrastructure or centralized administration. These nodes can be arbitrarily located and are free to move randomly at any given time, thus allowing network topology and interconnections between nodes to change rapidly and unpredictably. MANET is likely to be use in many practical applications, including personal area networks, home area networking, and military environments, and so on recent advances in wireless technology have enhanced the feasibility and functionality of wireless mobile ad hoc networks (MANETs). There has been significant research activity over the past 10 years into performance of such networks with the view to develop more efficient and robust routing protocols. However, there is majority research has concentrated on proactive or reactive routing protocol for data transmission, improving performance metrics and on the Security threats of this protocol by making change in it. But proactive and reactive both have some disadvantage as hybrid routing protocol come into existence is combination of both proactive and reactive, ZRP one among them come in to existence. Our contributions are as follows: Section I, introduces ZRP protocol and its component Section II, give details of previous and related work. In section III, we discuss about the simulation environment, in section IV, we discuss the result and in Section V, we conclude all the work and future work.

A. Zrp (Zone Routing Protocol)

ZRP [6] is a framework by using it we can take advantage of both table driven and on demand driven protocol according to the application. In this separation of nodes, local neighborhood from the global topology of the entire network allows for applying different approaches and thus taking advantage of each technique's features for a given situation. These local neighborhoods are called zones (hence the name) each node may be within multiple overlapping zones, and each zone may be of a different size. The "size" of a zone is not determined by geographical measurement, as one might expect, but is given by a radius of length α where α is the number of hops to the perimeter of the zone.

In the above diagram ZRP, protocol having Zone radius 2 in this in side the zone communication done in proactive way and out side it between such zones in reactive way. A, E, F, H, J, C are interior node and D, G, I, k are border nodes communication between B and K is done through proactive way and L is located out side the zone. ZRP consist of [8] three parts IARP [9] proactive part, IERP [10] reactive part of it and BRP [11] used with IERP to reduce the query traffic.

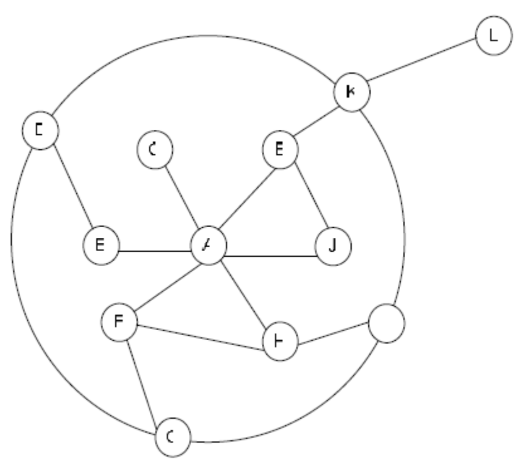

Figure 1 ZRP having Zone radius α =2

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B. Iarp (Intra Zone Routing Protocol)

The Intra zone Routing Protocol (IARP) [9] is a limited scope proactive routing protocol, which used to support a primary global routing protocol. The routing zone radius shows the scope of the proactive part, the distance in hops that IARP route updates relayed. lARP's proactive tracking of local network connectivity provides support for route acquiring and route maintenance. First, routes to local nodes are immediately available, avoiding the traffic overhead and latency of a route discovery. Traditional proactive link state protocols modified to serve as an IARP by limiting link state updates to the scope of the link source's routing zone.

C. Ierp (Inter Zone Routing Protocol)

The Interzone Routing Protocol (IERP) is the global reactive routing component of the Zone Routing Protocol (ZRP)[6]. IERP adapts existing reactive routing protocol implementations to take advantage of the known topology of each node's surrounding R-hop neighborhood (routing zone), provided by the Interzone outing Protocol (IARP) [9]. The availability of routing zone routes allows IERP to suppress route queries for local destinations. When a global route discovery is required, the routing zone based border cast service [11] used for efficiently guide route queries outward, rather than blindly relaying queries from neighbor to neighbor. Once a route discovered, IERP can use routing zones automatically to redirect data around failed links similarly, suboptimal route segments identified and traffic re-routed along shorter paths.

D. Brp (Bordercast Resolution Protocol)

The Bordercast Resolution Protocol (BRP) [11] provides the bordercasting packet delivery service. The BRP uses a map of an extended routing zone, provided by the local proactive Intrazone Routing Protocol (IARP) [9], to construct Bordercast (multicast) trees along which query packets are directed. (Within the context of the hybrid ZRP, the BRP used to guide the route requests of the global reactive Interzone Routing Protocol (IERP) [10]). The BRP uses special query control mechanisms to steer route requests away from areas of the network that have already covered by the query. The combination of multicasting and zone based query control makes Bordercasting an efficient and tunable service that is more suitable than flood searching for network probing applications like route discovery.
The Bordercast Resolution Protocol (BRP) is a packet delivery service, not a full featured routing protocol. Bordercasting enabled by local proactive Intrazone Routing Protocol (IARP) and supports global reactive Interzone Routing Protocol (IERP).

II. Related works

Nicklas Beijar in 2001 [5] first discuss the problem in proactive and reactive routing and then how they move towards the ZRP (Zone Routing Protocol) paper describe the architecture of the ZRP also describe the working of the protocol with an example. In 2002 Jan Schaumann [6] analyze the ZRP in mobile Adhoc network discuss the basic of MANET and implication on routing and problems occur due to rapidly changing topology without fixed router. In paper author, also discuss the ZRP hybrid routing protocol having both proactive and reactive protocol in context to other routing protocol. In 2003, David Oliver Jorg discusses the performance comparison of MANET routing protocol in different network size in that paper they discuss the problem due to the mobility of different nodes they test the routing performance of four different routing protocol. [7]in this examine the analytical simulation result for the routing protocol DSR ,TORA and ZRP emphasizing on the ZRP and impact of some of it most important attributes to the network performance. Julian Hsu, Sameer Bhatia, Mineo Takai, Rajive Bagrodia,[13] discuss the performance of common MANET routing protocol under realistic scenarios protocols include AODV OSPFv2 and ZRP which comprise all proactive, reactive ,hybrid routing protocol. In [14] discuss some of the factor that affects the routing algorithm like such as variable wireless link quality, propagation path loss, fading; multi-user interference, power expended and topological changes become important issues.. In paper, discuss about the proactive DSDV, WRP, CGSR, reactive SSR, AODV, RDMAR, Hybrid routing protocol like, ZRP. In [15] paper presents the idea of integrating the layer-II label-switching technique with layer-Ill and study the effect of Multiprotocol Label Switch (MPLS) mechanism on the performance Ad-Hoc Networks (MANETs). In 2007 [16] discuss the performance of three routing protocol DSR, AODV, LAR1 the performance is analyzed using varying, mobility and network size perform simulation on GLOMOSIM network simulator.

III. SIMULATION ENVIRONMENT

The simulation work done on Qualnet wireless network simulator version 4.0. Mobility model used is Random Way Point (RWP). In this model a Mobile node is initially placed in a random location in the simulation area, and then moved in an anomaly chosen direction between [0, 2] at a random speed between [SpeedMin, SpeedMax]. The movement proceeds for a specific amount of time or distance, and the process is repeated a predetermined number of times. We chose Min speed = 0 m/s, Max speed = 10 m/s, and pause time = vary. All the simulation work was carried out using ZRP routing protocol. Using Constant Bit Rate (CBR) sources provides network traffic. A CBR traffic source provides a constant stream of packets throughout the whole simulation thus further stressing the routing task.

A. Parameter Value For Simulation

- Mobility model Random Wave Point
  Minimum speed 0 mp
  Maximum speed 10 mps
- Pause time 30s
- Simulation Time 120s
  Terrain
    Coordination 800 * 800 m
    Connection

CBR (Constant Bit Ratio) Item size512(byte)

Radio/physical layer parameters Radio type: 802.lib Data rate: 2Mbps

B.3.2 Efficiency Metrics Used

- *Throughput*: It is the measure of the number of packets successfully transmitted to their final destination per unit time. It is the ratio between the numbers of sent packets vs. received packets.
- *Avg End to END Delay*: It signifies the average time taken by packets to reach one end to another end (Source to Destination).
- *Avg Jitter Effect:* It signifies the Packets from the source will reach the destination with different delays. A packet's delay varies with its position in the queues of the routers along the path between source and destination and this position can vary unpredictably.
- *Packet Loss Percentage:* It is the Ratio of transmitted packets that may have been discarded or lost in the network to the total number of packet sent.

IV. Results

Figure2 depicted that throughput of the ZRP having smaller zone radius decreases as compared to ZRP having higher zone radius as the node density increases. The possible reasons are as node density increases number of neighbor around the node increases and number of zones in the area increases. Due to this number of zones increases, so that reactive traffic of ZRP increases as compared to proactive one and large number of query packet are generated, to share information between zones.

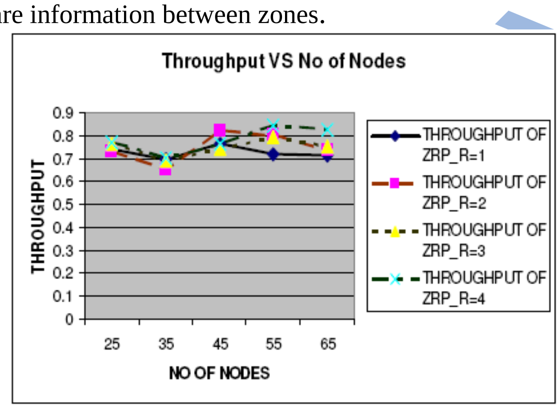

Figure 2 Comparison of throughput of ZRP in different node density by varying Zone radius.

Hence, large numbers of query packets are generated so chances of wrong path selection and time required for searching the destination increases. However, on the other side throughput of ZRP, having higher zone radius is better then the ZRP having smaller zone radius as the node density increases. The possible reasons are as the zone radius is increased zone size also increases and proactive traffic in ZRP increases as compared to reactive. Therefore, nodes have details of large number of neighbor around them, chance of query packet, data packet loss is less, and time required to share information with global part is decreases. As above, all discussion shows that ZRP having higher zone radius give the better throughput as compared to ZRP having smaller zone radius in high-density nodes.

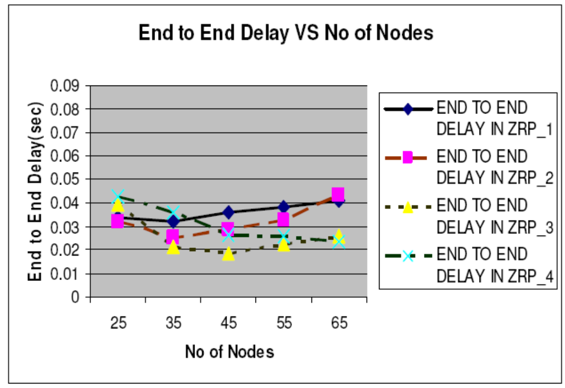

Figure 3 Comparison of End to End delay of ZRP in different node density by varying Zone radius.

Figure3 depicted that when the node density is less, ZRP having all zone radius almost give same end-to-end delay. The possible reason for this is as the density of node is less; number of neighbor around the node is less. Therefore, less number of update messages is required to take the details of nodes and time required to share information to exterior part reduced. Hence, overall delay required by the packet to reach destination from the source is almost same for all zone radius in ZRP.

However on the other side when node density increases end-to-end delay increases, in the ZRP having smaller zone radius as compared to ZRP having higher zone radius. The possible reason for this is as zone radius is smaller, number of zone increases. Due to this reactive traffic increases and chance of query, packet loss is also more and time required to share information between zone increases. Therefore, due to all these overall time delay required by the packet to reach the destination form the source increases. On the other hand, ZRP having higher zone radius shows less end-to-end delay as compared to ZRP having smaller zone radius. The possible reason is as the zone radius increases zone size also increases and proactive traffic of the ZRP used more as compared to reactive. Hence details of large number of node is available, so less time is required to share the information with global part, because of all this over all time delay taken by the packet to reach destination form source is reduced. Above all discussion shows that ZRP having higher zone radius produce less end-to-end delay as compared to ZRP having smaller zone radius in high-density node.

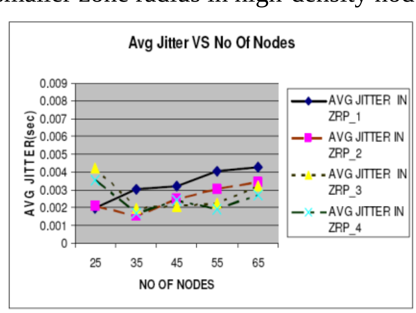

Figure 4 Comparison of Avg jitter Effect of ZRP in different node density by varying Zone Radius.

Figure 4 depicted that Avg jitter effect increases in ZRP having smaller zone radius as compared to ZRP having higher zone radius when node density increases. The possible reason for this is as zone radius is small number of zone increases, and reactive traffic in the ZRP increases as compared to proactive. Therefore, large numbers of query packet generated to search the path between zones. In these chances of query packet loss increases, hence time required for sharing information between zones vary, because of this packet form source reach the destination at different time delay.

However, on the other side Avg jitter effect is less in ZRP having higher zone radius as compared to ZRP having smaller zone radius in high-density node. The possible explanation is as the zone radius increases zone size also increases and number of zone reduced. Due to this proactive traffic in ZRP is more as compared to reactive traffic. Therefore, a detail of large number of nodes is available so chances of query packet loss are less. Due to this time required sharing, information with global part reduced and packet form the source to destination reach at equal interval.

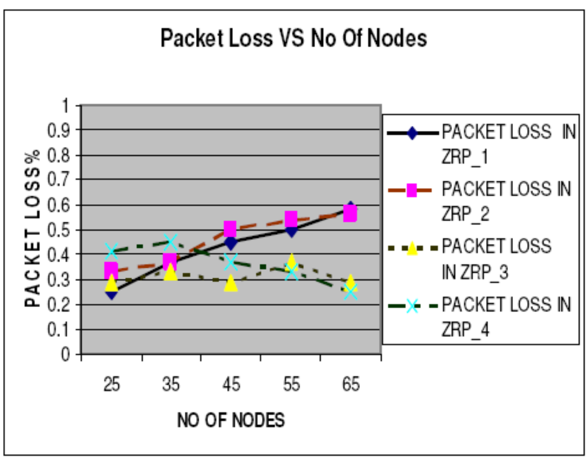

Figure 5 Comparison of packet loss using ZRP in different node density by varying Zone Radius.

Figure 5 depicted that when node density increases packet loss increases in ZRP smaller zone radius, as compared ZRP having higher zone radius. The possible explanation is as the node density high number of neighbor around the node increases, and number of zone increases. Due to this reactive traffic in ZRP, is more as compared to proactive. Therefore, a chance of query packet, data packet loss and wrong path selection increases.

However, on the other side as node density increases packet loss is less in ZRP having high zone radius as compared to ZRP having smaller zone radius. The possible explanation is as the node density high number of neighbor around the node also increases. Moreover, if zone radius is higher zone size increases, and number of zones decreases. Hence, proactive traffic in ZRP is more as compared to reactive and zone size is large so details of larger number of nodes are available. The reactive part is less, chances of query packet loss and packet loss due to wrong path selection also reduced.

V. Conclusions and Future Work

Node density has truly shown the effect on the performance of the ZRP protocol. As the density, changes ZRP attribute Zone radius has to be changed to get good performance. Result shows that configuration of Zone radius according to what type of application in which we use ZRP protocol. The high-density increases may increase the discovered services but it deteriorates their quality in terms of availability. If it is used for real time application likes video transmission then due to jitter effect performance decreases. In other application in which delay is consider then we can use the reduced Zone radius. Because as we increase the proactive part by increasing the Zone radius control traffic also increases. ZRP is suitable for the large network by providing the benefit of both proactive and reactive routing protocol.

As part of our future work we simulate ZRP by varying mobility and check its performance. Also check the performance of ZRP without using BRP it is interesting to see the performance of ZRP in large and realistic scenario.

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- Left Margin: 0.65
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- Font type of all text should be Times New Roman.
- Paper Title should be of Font Size 24 with one Column section.
- Author Name in Font Size of 11 with one column as of Title.
- Abstract Font size of 9 Bold, “Abstract” word in Italic Bold.
- Main Text: Font size 10 with justified two columns section
- Two Column with Equal Column with of 3.38 and Gaping of .2
- First Character must be two lines Drop capped.
- Paragraph before Spacing of 1 pt and After of 0 pt.
- Line Spacing of 1 pt
- Large Images must be in One Column
- Numbering of First Main Headings (Heading 1) must be in Roman Letters, Capital Letter, and Font Size of 10.
- Numbering of Second Main Headings (Heading 2) must be in Alphabets, Italic, and Font Size of 10.

You can use your own standard format also.

Author Guidelines:

1. General,

2. Ethical Guidelines,

3. Submission of Manuscripts,

4. Manuscript’s Category,

5. Structure and Format of Manuscript,

6. After Acceptance.

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3) Final approval of the version of the paper to be published.

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Papers: These are reports of significant research (typically less than 7000 words equivalent, including tables, figures, references), and comprise:

(a)Title should be relevant and commensurate with the theme of the paper.

(b) A brief Summary, "Abstract" (less than 150 words) containing the major results and conclusions.

(c) Up to ten keywords, that precisely identifies the paper's subject, purpose, and focus.

(d) An Introduction, giving necessary background excluding subheadings; objectives must be clearly declared.

(e) Resources and techniques with sufficient complete experimental details (wherever possible by reference) to permit repetition; sources of information must be given and numerical methods must be specified by reference, unless non-standard.

(f) Results should be presented concisely, by well-designed tables and/or figures; the same data may not be used in both; suitable statistical data should be given. All data must be obtained with attention to numerical detail in the planning stage. As reproduced design has been recognized to be important to experiments for a considerable time, the Editor has decided that any paper that appears not to have adequate numerical treatments of the data will be returned un-refereed;

(g) Discussion should cover the implications and consequences, not just recapitulating the results; conclusions should be summarizing.

(h) Brief Acknowledgements.

(i) References in the proper form.

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References

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Informal Tips for writing a Computer Science Research Paper to increase readability and citation

Before start writing a good quality Computer Science Research Paper, let us first understand what is Computer Science Research Paper? So, Computer Science Research Paper is the paper which is written by professionals or scientists who are associated to Computer Science and Information Technology, or doing research study in these areas. If you are novel to this field then you can consult about this field from your supervisor or guide.

Techniques for writing a good quality Computer Science Research Paper:

1. Choosing the topic- In most cases, the topic is searched by the interest of author but it can be also suggested by the guides. You can have several topics and then you can judge that in which topic or subject you are finding yourself most comfortable. This can be done by asking several questions to yourself, like Will I be able to carry our search in this area? Will I find all necessary recourses to accomplish the search? Will I be able to find all information in this field area? If the answer of these types of questions will be "Yes" then you can choose that topic. In most of the cases, you may have to conduct the surveys and have to visit several places because this field is related to Computer Science and Information Technology. Also, you may have to do a lot of work to find all rise and falls regarding the various data of that subject. Sometimes, detailed information plays a vital role, instead of short information.

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26. Go for seminars: Attend seminars if the topic is relevant to your research area. Utilize all your resources.

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Key points to remember:

- Submit all work in its final form.
- Write your paper in the form, which is presented in the guidelines using the template.
- Please note the criterion for grading the final paper by peer-reviewers.

Final Points:

A purpose of organizing a research paper is to let people to interpret your effort selectively. The journal requires the following sections, submitted in the order listed, each section to start on a new page.

The introduction will be compiled from reference matter and will reflect the design processes or outline of basis that direct you to make study. As you will carry out the process of study, the method and process section will be constructed as like that. The result segment will show related statistics in nearly sequential order and will direct the reviewers next to the similar intellectual paths throughout the data that you took to carry out your study. The discussion section will provide understanding of the data and projections as to the implication of the results. The use of good quality references all through the paper will give the effort trustworthiness by representing an alertness of prior workings.

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An abstract is a brief distinct paragraph summary of finished work or work in development. In a minute or less a reviewer can be taught the foundation behind the study, common approach to the problem, relevant results, and significant conclusions or new questions.

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- What you account in an conceptual must be regular with what you reported in the manuscript
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- Explain the value (significance) of the study
- Shield the model - why did you employ this particular system or method? What is its compensation? You strength remark on its appropriateness from a abstract point of vision as well as point out sensible reasons for using it.
- Present a justification. Status your particular theory (es) or aim(s), and describe the logic that led you to choose them.
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- Sort out your thoughts; manufacture one key point with every section. If you make the four points listed above, you will need a least of four paragraphs.
- Present surroundings information only as desirable in order hold up a situation. The reviewer does not desire to read the whole thing you know about a topic.
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Materials:

- Explain materials individually only if the study is so complex that it saves liberty this way.
- Embrace particular materials, and any tools or provisions that are not frequently found in laboratories.
- Do not take in frequently found.
- If use of a definite type of tools.
- Materials may be reported in a part section or else they may be recognized along with your measures.

Methods:

- Report the method (not particulars of each process that engaged the same methodology)

- Describe the method entirely
- To be succinct, present methods under headings dedicated to specific dealings or groups of measures
- Simplify - details how procedures were completed not how they were exclusively performed on a particular day.
- If well known procedures were used, account the procedure by name, possibly with reference, and that's all.

Approach:

- It is embarrassed or not possible to use vigorous voice when documenting methods with no using first person, which would focus the reviewer's interest on the researcher rather than the job. As a result when script up the methods most authors use third person passive voice.
- Use standard style in this and in every other part of the paper - avoid familiar lists, and use full sentences.

What to keep away from

- Resources and methods are not a set of information.
- Skip all descriptive information and surroundings - save it for the argument.
- Leave out information that is immaterial to a third party.

Results:

The principle of a results segment is to present and demonstrate your conclusion. Create this part a entirely objective details of the outcome, and save all understanding for the discussion.

The page length of this segment is set by the sum and types of data to be reported. Carry on to be to the point, by means of statistics and tables, if suitable, to present consequences most efficiently.

You must obviously differentiate material that would usually be incorporated in a study editorial from any unprocessed data or additional appendix matter that would not be available. In fact, such matter should not be submitted at all except requested by the instructor.
Content

- Sum up your conclusion in text and demonstrate them, if suitable, with figures and tables.
- In manuscript, explain each of your consequences, point the reader to remarks that are most appropriate.
- Present a background, such as by describing the question that was addressed by creation an exacting study.
- Explain results of control experiments and comprise remarks that are not accessible in a prescribed figure or table, if appropriate.
- Examine your data, then prepare the analyzed (transformed) data in the form of a figure (graph), table, or in manuscript form.

What to stay away from

- Do not discuss or infer your outcome, report surroundings information, or try to explain anything.
- Not at all, take in raw data or intermediate calculations in a research manuscript.
- Do not present the similar data more than once.
- Manuscript should complement any figures or tables, not duplicate the identical information.
- Never confuse figures with tables - there is a difference.

Approach

- As forever, use past tense when you submit to your results, and put the whole thing in a reasonable order.
- Put figures and tables, appropriately numbered, in order at the end of the report
- If you desire, you may place your figures and tables properly within the text of your results part.

Figures and tables

- If you put figures and tables at the end of the details, make certain that they are visibly distinguished from any attach appendix materials, such as raw facts
- Despite of position, each figure must be numbered one after the other and complete with subtitle
- In spite of position, each table must be titled, numbered one after the other and complete with heading
- All figure and table must be adequately complete that it could situate on its own, divide from text

Discussion:

The Discussion is expected the trickiest segment to write and describe. A lot of papers submitted for journal are discarded based on problems with the Discussion. There is no head of state for how long a argument should be. Position your understanding of the outcome visibly to lead the reviewer through your conclusions, and then finish the paper with a summing up of the implication of the study. The purpose here is to offer an understanding of your results and hold up for all of your conclusions, using facts from your research and generally accepted information, if suitable. The implication of result should be visibly described.
Infer your data in the conversation in suitable depth. This means that when you clarify an observable fact you must explain mechanisms that may account for the observation. If your results vary from your prospect, make clear why that may have happened. If your results agree, then explain the theory that the proof supported. It is never suitable to just state that the data approved with prospect, and let it drop at that.

- Make a decision if each premise is supported, discarded, or if you cannot make a conclusion with assurance. Do not just dismiss a study or part of a study as "uncertain."
- Research papers are not acknowledged if the work is imperfect. Draw what conclusions you can based upon the results that you have, and take care of the study as a finished work
- You may propose future guidelines, such as how the experiment might be personalized to accomplish a new idea.
- Give details all of your remarks as much as possible, focus on mechanisms.
- Make a decision if the tentative design sufficiently addressed the theory, and whether or not it was correctly restricted.
- Try to present substitute explanations if sensible alternatives be present.
- One research will not counter an overall question, so maintain the large picture in mind, where do you go next? The best studies unlock new avenues of study. What questions remain?
- Recommendations for detailed papers will offer supplementary suggestions.

Approach:

- When you refer to information, differentiate data generated by your own studies from available information
- Submit to work done by specific persons (including you) in past tense.
- Submit to generally acknowledged facts and main beliefs in present tense.

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	A-B	**C-D**	**E-F**
Abstract	Clear and concise with appropriate content, Correct format. 200 words or below	Unclear summary and no specific data, Incorrect form Above 200 words	No specific data with ambiguous information Above 250 words
Introduction	Containing all background details with clear goal and appropriate details, flow specification, no grammar and spelling mistake, well organized sentence and paragraph, reference cited	Unclear and confusing data, appropriate format, grammar and spelling errors with unorganized matter	Out of place depth and content, hazy format
Methods and Procedures	Clear and to the point with well arranged paragraph, precision and accuracy of facts and figures, well organized subheads	Difficult to comprehend with embarrassed text, too much explanation but completed	Incorrect and unorganized structure with hazy meaning
Result	Well organized, Clear and specific, Correct units with precision, correct data, well structuring of paragraph, no grammar and spelling mistake	Complete and embarrassed text, difficult to comprehend	Irregular format with wrong facts and figures
Discussion	Well organized, meaningful specification, sound conclusion, logical and concise explanation, highly structured paragraph reference cited	Wordy, unclear conclusion, spurious	Conclusion is not cited, unorganized, difficult to comprehend
References	Complete and correct format, well organized	Beside the point, Incomplete	Wrong format and structuring

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