Spring 2-17-2013

International Conference On Electronics Engineering and Computer Science ICEECS-2013

Prof. Dr.Srikanta Patnaik

Editorial

Electronics Engineering and Computer Science deals with recent developments and practices adopted in various projects in different Engineering disciplines and specializations .Traditionally computing studies occupy two partitions, (Science and Engineering) separated by a line roughly at the Computer Architecture level. A more effective organization for Computer Science and Engineering requires an intrinsically interdisciplinary framework that combines academic and systems-oriented computing perspectives. Researchers have been developing such a framework, which aggregates Computer Science and Computer Engineering, then repartitions the resulting single field into analysis and synthesis components. The framework is based on the notion that Science is foremost about dissecting and understanding and Engineering is mostly about envisioning and building. The computer had a great effect on Communication. Electronics Engineering and Computer Science runs together with hand in hand.

The idea of modeling in a computer and with the aid of a computer is under trial. For modeling, Computer is the basic Infrastructure to centralize communication. Any communication between people about the same concept is a common revelatory experience about informational models of that concept. Each model is a conceptual structure of abstractions formulated initially in the mind of one, and while communicating if it is different from those in the mind of other, there is no common model and no communication. Researchers are working on applying their wireless and mobile research to transportation, health care, education, collaboration and environmental sustainability. Projects already underway include safe and efficient road transportation, autonomous driving, wireless medical implants, mobile video delivery, multiparty wireless videoconferencing and energy harvesting.

The Conference sometimes is conducted in collaboration with other Institutions.IRNet encourages and invite proposals from Institutes within India and Abroad to join hands to promote research in various areas of discipline. These conferences have not only promoted the international exchange and cooperation, but have also won favorable comments from National and International participants, thus enabled IRNet to reach out to a Global network within three years time. The conference is first of its kind and gets granted with lot of blessings.

The Conference designed to stimulate the young minds including Research Scholars, Academicians, and Practitioners to contribute their ideas, thoughts and nobility in these disciplines of Engineering. IRNet received a great response from all parts of country and Abroad for the presentation and publication in the proceeding of the conference.

I sincerely thank all the authors for their invaluable contribution to this Conference. I am indebted towards the Reviewers and Board of Editors for their generous gifts of time, energy and effort. It’s my pleasure to welcome all the participants, delegates and organizer to this International Conference on behalf of IRNet family members. IRNet believes to make ***“RESEARCH COOL”.***

I wish all success to the paper presenters. The papers qualifying the review process will be published in the forthcoming IOAJ journal.

Convenor:-

Mr.Rashmi Ranjan Nath

IRNet, INDIA,

Research Associate

AUTOMATED VOICE BASED HOME NAVIGATION SYSTEM FOR QUADRIPLEGIC PATIENTS

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Abstract- In this paper, we propose an Intelligent Home Navigation System (IHNS) which comprises of a wheelchair, voice module and navigation module. It can be used by an elderly or physically challenged person to move inside the home without any difficulty. The voice of the person is detected by voice capture module which will be compared by voice recognition module with predefined voices loaded in to the system. According to the received voice, the direction is automatically understood and the wheelchair moves according to the route which is predefined. It is also equipped with obstacle avoidance technique, where the person may not be able to provide proper voices at the right time. The wheel chair can automatically navigate from one point to the other in the home as per predefined route based on the voice received. Thus the above proposed system can be used by elderly and physically challenged people in day to day life even if they are alone at home. This wheelchair is basically used by quadriplegic patients who cannot move their body parts but can move their head completely or upto certain extent and can also speak clearly.

Keywords- *wheelchair, elderly, physically challenged, voice, obstacle avoidance, quadriplegic.*

I. INTRODUCTION

The most common image of disability is the people in Wheelchairs. Wheelchairs are used by people who find themselves unequipped to move without external aid. The special needs of the elderly may differ from that of a physically challenged person or a large individual but they all have "special needs" and often require some assistance to perform their daily routine. The required assistance may be due to ageing, physical limitations, medical conditions or other issues. The physically challenged people who use a normal wheelchair for navigation, usually requires an external person to move around. In this busy world, the elderly people may be left alone at home and also may not find an apt person for external help. Here comes the need of an automated home navigation system, which consists of a wheelchair which can be used by the elderly and the physically challenged people without the help of an external person. The proposed system can be operated using voices which is coded into it. This problem is also dealt in this system as it navigates automatically. Another important feature is that the personal security of the person who is using the wheelchair is also taken care. If the person feels uncomfortable or insecure, he can avail the emergency service like police or hospital by making use of the predefined voices to it.

II. MOTIVATION

Today's world comprises of a large variety of people. Some of them depend on others for their living. But in today's fast world, everyone is busy and there are less people to care for the increasing number of elderly and the physically challenged people. Also these people find it tough to even navigate inside the home without external aids. The elderly people find automated wheelchairs as an easy way for locomotion having known about these facts, our aim was to bring an automated navigation system which can be used by both – the elderly and the physically challenged people in a user-friendly manner using voices for operation.

III. PROBLEM DEFINITION

The elderly and the physically challenged people find it tough for locomotion without the help of external aid. Studies have shown that until quite recently disabled people were socially isolated. Whether their condition was physical, emotional or mental, all met the same attitudes. They were kept off from social gatherings because they needed special attention or people to take care of them. These miserable conditions made us think of bringing out a system which include personal security features and can be used by these misfortunate people so that they can navigate easily and without external aids.

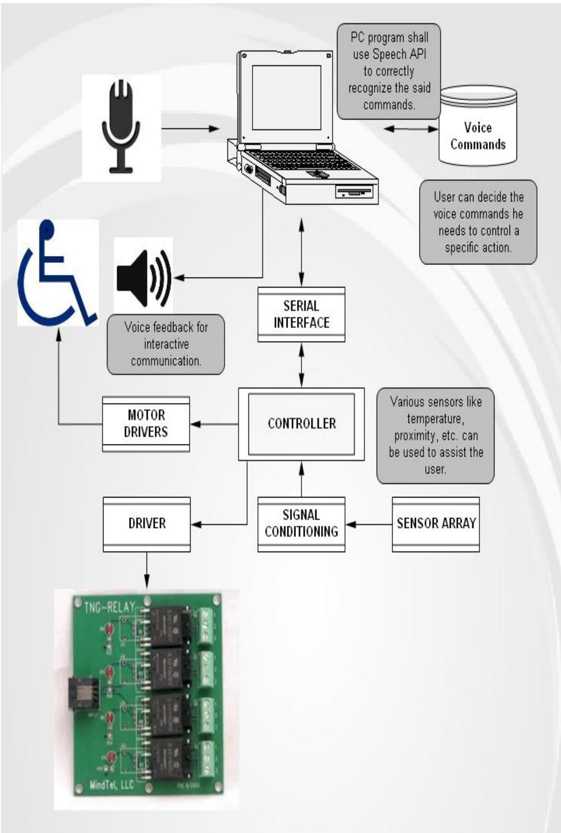
IV. PROJECT SCOPE

- In this project, we propose an Intelligent Home Navigation System which comprises of a wheelchair, voice module and navigation module. It can be used by an elderly or physically challenged person to move inside the home without any difficulty.
- This paper presents the design of a navigation system and its integration with a commercial powered wheelchair. The navigation system provides the commercial wheelchair with a set

of functions which increase the autonomy of elderly and people with motor disabilities.

- Our system comprises of integration of voice processing and embedded technology. Microcontroller unit is interface with the wheelchair. The voice of the patient is received by the microphone connected to the PC. In PC voice is processed and the signal is given to the microcontroller and the motor is driven accordingly.

V. SYSTEM ARCHITECTURE



VI. MATHEMATICAL MODEL

Mathematical model is described using motion detection algorithm. Motion detection algorithm is actually to compare the current frame with the previous one. It's possible to compare the current frame not with the previous one but with the first frame in the video sequence. So, if there were no objects in the initial frame, comparison of the current frame with the first one will give us the whole moving object independently of its motion speed. It's useful in video compression when you need to estimate changes and to write only the changes, not the whole frame.

1. S={Ir,Icur,Iseq}

Where,
Ir- registered input frame
Icur-current frame
Iseq-sequence of frames fed in the camera
Iseq={i1,i2,………..}

2. ST=f(Ir,Icur,threshold)

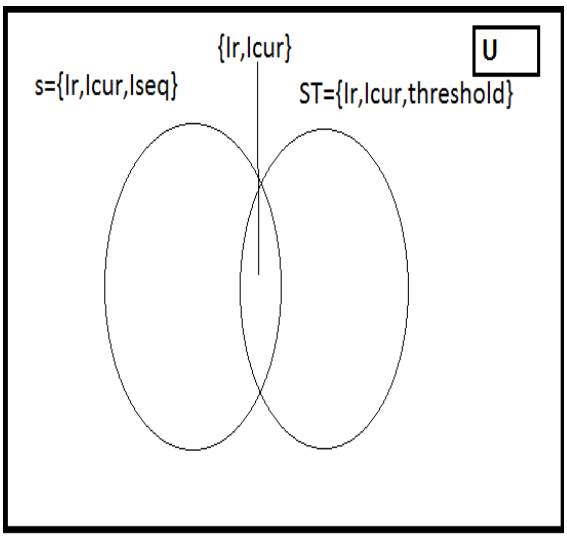
Where,
ST-status
f-function for comparison
threshold-fixed input value
Status is used to detect change in motion of the object with a particular fixed frame. The function compares all the three values. If it returns true value then change in motion of the object is detected.

Psuedo code:

```
P1=co-ordinates of registered input
P2=co-ordinates of current image
H=height of the object
W=width of the object
Initialise count=0
For(y=0 to h)
{
 For (x=0 to w)
 {
 P1=i1[x][y];
P2=i2[x][y];
B1=brightness(p1);
B2=brightness(p2);
Diff=abs(B1-B2);
If(diff>threshold)
    {
       Count++;
     }
  }
}
Int per=(count/w*h)*100;


If(per>threshold)
{
 Alarm=active;
}
```

VII. SET THEORY



Our project automated vaice based home navigation system consists of two sets S and ST.
S={Ir,Icur,Iseq}
Ir->it is the registered frame which is already registered in the database.
Icur->it is the current frame which is to be compared.
Iseq->it is the set of incoming frames.
Iseq={i1,i2,i3,……….}

ST={Ir,Icur,threshold}
Threshold=it is the fixed value.the comparision is to be done with this value.
Here,

- S is the input set.
- ST is the output set

S={Ir,Icur,Iseq}---------------------→initial state.
ST={Ir,Icur,threshold}----------------→final state.

ADVANTAGES

- Highly beneficial for quadriplegic patient.
- Highly portable.
- Provides required assistance to elderly and physically challenged
- Personal security provided by alarms.
- Automation reduces dependency on others.

DISADVANTAGES

- Costly
- Totally helpless when obstacle is present on all the 4 sides.
- Battery dependent.

VIII. CONCLUSION

- In this paper we have discussed about an automated voice based navigation system which can be used by anyone who requires the help of others for their day to day locomotion.
- This setup is very helpful mainly for elderly and physically challenged people and thus is a noble cause.
- As it is controlled using voices, it can be regarded as a very user-friendly system.
- It has some special features integrated into it like the personal security module and the obstacle avoidance techniques.
- This is very useful since the elders or the physically challenged needs medical aid at any time during the day.

REFERENCES

[1]. Special Needs and Daily living – Special needs Solutions for all.

[2]. Ase brandt, Susanne iwarsson and Agneta sta, "Older People’s Use of Powered Wheelchairs for Activity And Participation, Taylor and Francis health Sciences.

[3]. Ren C. Luo, Tse Min Chen, Chi-Yang Hu, and Zu Hung Hsiao, ‘Adaptive Intelligent Assistance Control of Electrical Wheelchairs by Grey-Fuzzy Decision-Making Algorithm’ , Proceedings of the 1999 IEEE International Conference on Robotics & Automation Detroit, Michigan May 1999.

[4]. Yoshinori Kunotl, Teruhisa Murashimat , Nobutaka Shimadat and Yoshiaki Shirait ‘Interactive Gesture Interface for Intelligent Wheelchairs’.

[5]. Donald P. Massa, ‘Choosing an Ultrasonic Sensor for Proximity or Distance Measurement Part 1: Acoustic Considerations’

[6]. S. Fioretti, T. Leo, and S. Longhi, ‘A Navigation System for Increasing the Autonomy and the Security of Powered Wheelchairs’, IEEE Transactions On Rehabilitation Engineering, Vol. 8, No. 4, December 2000

[7]. Pei Jia, Huosheng H Hu, Tao Lu Kui Yuan, ‘Head gesture recognition for hands-free control of an intelligent wheelchair’,

[8]. Yoshifumi Murakamit , Yoshinori Kuno, Nobutaka Shimadat and Yoshia, ‘Collision Avoidance by Observing Pedestrians Faces for Intelligent Wheelchairs’, Proceedings of the 2001 IEFWRSJ International Confermm on Intelligent Robots Mwi,Hawaii, USA, Oct. 29-NOV0.3,2009. ISBN.

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DYNAMIC PATH UPDATE (DPU) BASED HOP COUNT FILTERING FOR SENSING AND STAMP-OUT OF IP SPOOFING

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Abstract- IP Spoofing has often been exploited by Distributed Denial of Service (DDoS) attacks to (1) conceal flooding sources and dilute localities in flooding traffic, and (2) coax legitimate hosts into becoming reflectors, redirecting and amplifying flooding traffic. Thus, the ability to filter spoofed IP packets near victim servers is essential to their own protection and prevention of becoming involuntary DoS reflectors. Although an attacker can forge any field in the IP header, he cannot falsify the number of hops an IP packet takes to reach its destination. More importantly, since the hop-count values are diverse, an attacker cannot *randomly* spoof IP addresses while maintaining consistent hop-counts. On the other hand, an Internet server can easily infer the hop-count information from the Time-to-Live (TTL) field of the IP header. Using IP to Hop Count mapping, the server can distinguish spoofed IP packets from legitimate ones. Based on this observation, we present a novel filtering technique, called **Dynamic Path Update based HCF** that builds Dynamic IP to Hope Count mapping table—to detect and discard spoofed IP packets. Dynamic Path Update based HCF is easy to deploy, as it does not require any support from the underlying network.

Keywords: *IP Spoofing, DDoS, HCF*

I. INTRODUCTION

Network Security is one of the main domains of Information Technology and DDoS is one of the main threats to the Network Security. DDoS attacks servers and serves as hindrance to various security policies. The IP spoofing is one of the advanced methods of the DDoS attacks. Distributed Denial of services (DDoS) attacks is virulent, relatively new type of attack on the availability of internet service and resources [8]. DDoS attacker infiltrates large number of computers by exploiting software vulnerabilities, to setup DDoS attack networks. These unwitting computers are then invoking to wage a coordinator, large scale attack against one or more victims systems. As specific countermeasures are developed, attackers enhanced existing DDoS attack tools, deriving new techniques [9]. Hence, it would be desirable to develop comprehensive DDoS solution that defend against known and futures DDoS attacks variant. In 2000, there was severe attack on high profile website such has yahoo.com, CNN.com, amazon.com. In 2002, 8 out of 13 root DNS server were brought down has result of severe flooding denial of service attack [10]. Some proposal tries to detect spoofed senders using new routing mechanism such as "path markers supported by some or the entire router in root, as in Pi [11]. Few proposal try to detect spoofed senders using existing mechanisms, such as hop count (time to live)), as in Hop Count Filtering (HCF) [12]. However, empirical evaluation of these approaches show rather disappointing results [13].

The Source Router Preferential Dropping (SRPD) is proposed in [14]. The SRPD scheme monitors incoming high rate flows and preferentially dropped their packet. The dropping decision is based on flow rate threshold violation, the victim server's response time, and the victim router queue occupancy. The IP trace back mechanism is one such approach to identify the hosts which were involved in an attack [15]–[19]. In a marking scheme [10], packets are marked probabilistically by intermediate routers, hence facilitating the victim network to identify the path traversed by the attack packets. A similar scheme, Tabu Marking Scheme (TMS) is proposed in [15].

(Hop Count Filtering (HCF) is one of the best methods to overcome IP spoofing. The IP protocol lacks the control to prevent a sender from hiding its packets' origin. Moreover, routers don't store IP address of each packet. Hence sender's identity is not known by the routers. The TTL which is stored in each packet is accessed by the HCF so that it may be used to validate them and hence it does not need to trace back

Based on hop-count, we propose a novel filtering technique, called ***Dynamic Path Update based Hop-Count Filtering*** (DPU based HCF), to weed out spoofed IP packets at the very beginning of network processing, thus effectively protecting victim servers' resources from abuse. The rationale behind HCF is that most randomly-spoofed IP packets, when arriving at victims, do not carry hop-count values that are consistent with the IP addresses being spoofed. As a receiver, an Internet server can infer the hop-count information and check for consistency of source IP addresses. In existing system (HCF), receiver check only unique path between source and destination that is specified in the IP2HC table. Hence the condition not satisfied, it will discard the packet. In our proposed system, dynamic path can be evaluated and updated with priorities in receiver table. During transmission time, if routers change the path, then the packet will be discarded in receiver, according to the existing system. Using our proposed system, when ever the router chooses alternate path, the receiver checks each prioritized hop count only when hop counts is satisfied, then it will regard as legitimate packet else packet will be discarded.

II. LITERATURE REVIEW

During the earlier days, DDoS attacks were employed for IP spoofing. This was overcome by ingress filtering [1] that detected spoofed packet but it was not much effective. Reflectors [2] can be used to protect against the distributed denial of service attacks. Reflectors reflect the DDoS that have been sent by zombies or the hacker itself. These reflectors did not fulfill several hosts' expectation.

The study of previous proposals tries to detect spoofed senders using new rooting mechanism such as "path markers supported by some or the entire router in root, as in Pi [11]. The IP trace back mechanism is one such approach to identify the hosts which were involved in an attack [15]–[19]. In a marking scheme [10], packets are marked probabilistically by intermediate routers, hence facilitating the victim network to identify the path traversed by the attack packets.

TCP service which has been used to great extent has been affected by IP spoofing. It is protected by various methodologies [3].

There is a scheme proposed for flooding attacks detection however, our research focus is on both high-rate and IP-spoofing attacks. The Multilevel Tree for Online Packet Statistics (MULTOPS) [20] provides a data structure for DDoS attack detection. The basic idea is that during normal operation, the packet rate of traffic in one direction is proportional to the packet rate in the other direction.Jin et al. [21] proposed Hop-Count Filtering (RCF) for Internet servers to winnow away spoofed IP packets. The rationale behind RCF is that an attacker cannot alter the number of hops an IP packet takes to reach its destination, though he can forge any field in the IP header. The most randomly- spoofed IP packets, when arriving at victims, do not carry hop count values that are consistent with the IP addresses being spoofed. On the other hand, an Internet server can easily infer the hop count information from the TTL field of the IP header. Exploiting this observation, RCF builds an IP2RC mapping table to detect and discard spoofed IP packets, by clustering address prefixes based on hop counts. Using a mapping between IP addresses and their hop-counts, the server can distinguish spoofed IP packets from legitimate ones. Based on this observation, we present a novel filtering technique, called *Hop-Count Filtering* (HCF)—which builds an accurate IP-to-hop-count (IP2HC) mapping table—to detect and discard spoofed IP packets. HCF is easy to deploy, as it does not require any support from the underlying network. Through analysis using network measurement data, we show that HCF can identify close to 90%of spoofed IP packets, and then discard them with little collateral damage. We implement and evaluate HCF in the Linux kernel, demonstrating its effectiveness with experimental measurements.

III. PROPOSED SYSTEM

Dynamic Path Update based HCF which builds an all possibilities of IP-to-hop-count (IP2HC) mapping table—to detect and discard spoofed IP packets. DPU based HCF is easy to deploy, as it does not require any support from the underlying network. Through analysis using network measurement data, we show that DPU based HCF can identify more than 90% of spoofed IP packets, and then it check next possibilities (DYNAMIC) path to reach destination because there are many possibilities of In our system dynamic path can be evaluated and filled in receiver table. In case of alternate path during the traffic time, the alternate path were checked at the receiver table and packet is acceptable if it satisfies the condition else packet is discarded.

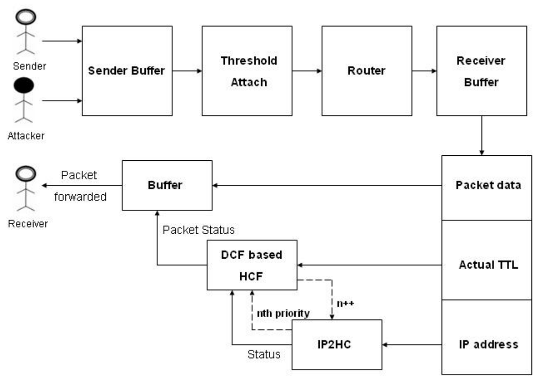

Fig 1: Overall System Design

routing path between source and destination. While the next path satisfies the condition then packet is forwarded to the receiver and update the HCF table else packet is discarded. In existing system (HCF), receiver check only accurate path between source and destination if it does not satisfied then packet is discarded. As per our proposed work, sender will have to first store the data in the sender buffer. Since attacker can easily evade the security barriers of the system, it also stores the data spoofing the sender's identity. So packet from the both, sender and attacker is attacked with experimental threshold (Te) and forwarded to the intermediate routers. It is then forwarded to receiver buffer. Here, each packet is separated into 3 fields. Data from packet is given to the Buffer. Actual TTL packet is extracted and forwarded to the DPU based HCF. The IP address from the packet is mapped with the IP2HC table to get the corresponding Hop Count (or Threshold) which has highest priority. When Te does not match with the corresponding Ta, then next highest priority Te is obtained. This is followed till the nth priority Te. The resultant status is given to the Buffer. The Buffer after analyzing the status accepts or discards the packet. This is shown in fig: 1. Sender should be initialized with the port number and IP address. The sender should be authenticated before allowing it to send the packet. Packet should be attached by sender along with its IP address and TTL field. With each packet the sender sends, the HCF obtained from its corresponding threshold time.

Host Name	TTL(microseconds)	Priority
System 1	3	1
System 2	800	1
System 2	900	2
System 2	1000	3
System 3	500	1
System 4	600	1
System 5	550	1
System 6	700	1
System 7	850	1
System 8	800	1
System 9	900	1

Fig 2: INITIAL HCF TABLE

Source IP	Protocol	Packet	Status	Sender Time(micro second)
192.168.6.61	TCP	hi	Forward	1335523149640
192.168.6.61	TCP	800	Forward	1335523149645
192.168.6.61	TCP	hello	Forward	1335523149650
192.168.6.61	TCP	800	Forward	1335523149655
192.168.6.61	TCP	world	Forward	1335523149660
192.168.6.61	TCP	800	Forward	1335523149665
192.168.6.61	TCP	hi	Forward	1335523149700
192.168.6.61	TCP	1000	Forward	1335523149705
192.168.6.61	TCP	hello	Forward	1335523149710
192.168.6.61	TCP	1000	Forward	1335523149715
192.168.6.61	TCP	world	Forward	1335523149720
192.168.6.61	TCP	1000	Forward	1335523149725

Fig 3: SENDER TABLE

Source IP	Protocol	Packet	Status	Sender Time(micro
192.168.6.61	TCP	hi	Reached	1335523149640
192.168.6.61	TCP	800	Reached	1335523149645
192.168.6.61	TCP	hello	Reached	1335523149650
192.168.6.61	TCP	800	Reached	1335523149655
192.168.6.61	TCP	world	Reached	1335523149660
192.168.6.61	TCP	800	Reached	1335523149665
192.168.6.61	TCP	hi	Discarded	1335523149700
192.168.6.61	TCP	1000	Discarded	1335523149705
192.168.6.61	TCP	hello	Discarded	1335523149710
192.168.6.61	TCP	1000	Discarded	1335523149715
192.168.6.61	TCP	world	Discarded	1335523149720
192.168.6.61	TCP	1000	Discarded	1335523149725

Fig 4: RECEIVER TABLE AFTER DPU BASED HCF CHECK

Host Name	TTL(Microseconds)	Priority
System 1	3	1
System 2	900	1
System 2	800	2
System 2	1000	3
System 3	500	1
System 4	600	1
System 5	550	1
System 6	700	1
System 7	850	1
System 8	800	1
System 9	900	1

Fig 5: UPDATED HCF TABLE

The packet along with its HCF (as given in the fig: 2) and IP field is fed into the sender buffer (as in fig: 3). Router receives each packet and forwards to the receiver host with minimal traffic life time. Each time the router receives the packet it attach its TTL field along with it. The router forwards each packet to the receiver system.

Receiver stores each packet in its buffer (as given in fig: 4). It extracts the IP and TTL field and forwards the IP address to IP2HC table. The information obtained from this module is again forwarded to receiver.

The TTL field forwarded by the receiver is accepted by the verification module. This TTL is checked with the TTL obtained from the IP2HC table. When the value is same, the packet is considered as legitimate or else, it is discarded.The updated IP2HC table is forwarded to all the system as they will be having an updated IP2HC table of the other hosts in the network.

IV. ADVANTAGES

- The Proposed System reduces the resending process.
- In transmission time, the legitimate user can use the alternative path.
- The Proposed System accepts the dynamic path during transmission.

V. CONCLUSION

In this paper, we present a hop-count-based filtering scheme that detects and discards spoofed IP packets to conserve system resources. Our scheme inspects the hop-count of incoming packets to validate their legitimacy. Using only a moderate amount of storage, DPU based HCF constructs a Dynamic IP2HC mapping table via IP address aggregation and hop-count clustering. A pollution-proof mechanism initializes and updates entries in the mapping table. We have known that HCF can remove more than 90% of spoofed traffic. Moreover, even if an attacker is aware of HCF, he cannot easily circumvent HCF. Though we have described some advancements in this paper using Dynamic Path Update (DPU) based HCF concepts, it may also need some future enhancements. Some of the future enhancements are:

- The packet size can be reduced so as to decrease the network traffic.
- Dynamic clustering algorithm can be altered so that the packet size will be minimized.

DPU Based Hop Count Filtering Algorithm

DPU Based Hop Count Filtering Algorithm:

Required: Updated IP2HC table

Assumption: Both Sender and Receiver knows the Updated IP2HC table.

Process

1.Sender sends the legitimate packet along with the T_e(experimental threshold) to the routers.

2.Routers forwards the packet to the receiver by dynamic path allocation.

3.Receiver receives the packet and extracts the T_e from the packet.

4.Checks the T_e with the T_a (Actual Threshold calculated from the TTL field in the packet).

5.If it matches, it is regarded as legitimate packet or else it is forwarded to dynamic Hop Count Check().

dynamic Hop Count Check()

6.In Dynamic hop count check, the IP address of the packet is extracted and mapped to IP2HC table. The corresponding Thresholds of all priorities are extracted from the table.

7.The T_e is compared with each prioritized thresholds T_{ep}[p] (obtained from IP2HC table).

```
While(Tep[p]!='\0')
  { if(Tep==Te)
      {S(accepted);
       top=p;
      }
    else
      {p++;
      }
```

8.After checking with all possibilities if the threshold matches, status S and current priority P_i is obtained.

9. IP2HC table is updated with current P_i as the top priority and broadcasted all the nodes in the network.

REFERENCES

[1]. **"Packet Filtering Based on Source Router Marking and Hop-Count",**Kashif Ali, Mohammad Zulkernine, and Hossam Hassanein, 32nd IEEE Conference on Local Computer Networks,2007.

[2]. **"A Pi2HC Mechanism against DDoS Attacks",** Guang Jin, Yuan Li, Huizhan Zhang, Jiangbo Qian, College of Information Science and Engineering, Ningbo University, Ningbo, China,

[3]. **"Mitigating DDoS attack and Saving Computational Time using a Probabilistic approach and HCF method"**, Biswa Ranjan Swain and Bibhudatta Sahoo, 2009 IEEE International Advance Computing Conference (IACC 2009) Patiala, India, 6-7 March 2009.

[4]. **"Hop Count Based Packet Processing Approach to Counter DDoS Attacks",** Bharathi KrishnaKumar* , P.Krishna Kumar , Prof. Dr. R. Sukanesh, 2010 International Conference on Recent Trends in Information, Telecommunication and Computing. "**Protecting tcp services from denial of service attacks**", H. Farhat, in LSAD '06: Proceedings of the 2006 SIGCOMM Workshop on large scale Attack Defense. New York, NY, USA: ACM Press, 2006.

[5]. "**Network ingress filtering: Defeating denial of service attacks which employ IP source address spoofing",** P. Ferguson and D. Senie, In RFC 2267, January 1998.

[6]. "**An analysis of using reflectors for distributed denial-of-service attacks**", V. Paxson. ACM Computer Communication Review, 31(3), July 2001.

[8]. **"DDoS Attacks and Defense Mechanisms Classification "**, Douligeris, C.Mitrokotsa, A, Proceedings of 3d IEEE International Symposium, 14-17 Dec 2003.

[9]. **"Distributed Denial of Service: Taxonomies ofAttacks, Tools, and Countermeasures"**, Spetch, Stephen M. Lee, Ruby B**,** Proceedings of the 17th International Conference on Parallel and Distributed Computing Systems, 2004 International Workshop on Security in Parallel and Distributed Systems,September 2004.

[10] **"Accurately Measuring Denial of Service in Simulation and Testbed Experiments",**Jelena Mirkovic, Alefiya Hussain, Sonia Fahmy,, Peter Reiher, and Roshan K. Thomas,IEEE Transactions On Dependable And Secure Computing, VOL. 6, NO. 2, April-June2009

[11] **"StackPi: New Packet Marking andFiltering Mechanisms for DDoS and IP Spoofing Defense",** Abraham Yaar, Adrian Perrig, Dawn Song IEEE J. Selected Areas in Comm., vol. 24, no. 10, Oct. 2006.

[12]. **"Defense Against Spoofed IP Traffic Using Hop-Count Filtering",** Haining Wang, Cheng Jin, and Kang G. Shin IEEE Transactions on Networking, vol.15.No.1,Feb 2007.

[13]. **"Collaborative Detection of DDoS attacks over multiple network domains",**Chen, Hwang, and ku, ,IEEE Transactions on parallel and distributed systems, tpds-0228-0806,2006.

[14] "**Proactive control of distributed denial of service attacks with source router preferential dropping,"**Y. Fan, H. Hassanein,and P. Martin, in Proceedings of 3rd ACS/IEEE International Conference on Computer Systems and Applications, 2005.

[15] **"Tabu marking scheme to speedup ip traceback",**M. Ma, *Computer Networks*, vol. 50, no. 18, pp. 3536–3549, 2006.

[16] **S. Savage, D. Wetherall, A. Karlin, and T. Anderson, "Practical network support for ip traceback**," *ACM SIGCOMM Computer Communication Review*, vol. 30, no. 4, pp. 295–306, 2000.

[17] **D. X. Song and A. Perrig, "Advanced and authenticated marking schemes for ip traceback**." in *IEEE INFOCOM*, 2001, pp. 878–886.

[18**]** **A. C. Snoeren, "Hash-based ip traceback,**" in *SIGCOMM '01: Proceedings of the 2001 Conference on Applications,Technologies, Architectures, and Protocols for Computer Communications*.New York, NY, USA: ACM Press, 2001, pp. 3–14.

[19] **R. stone, "Centertrack: An ip overlay network for tracking dos floods**," in *Proceedings of* 9*th USENIX Security Symposium*, 2000, pp. 199–212.

[20] **M. P. Thomer M. Gil, "Multops: a data-structure for bandwidth attack detection**," in *Proceedings of* 10*th USENIX security Symposium*, 2001.

[21] **C. Jin, H. Wang and K. G. Shin, "Hop- count Filtering: An Effective Defense Against Spoofed Traffic**," In Proceedings of the 10th ACM Conference on Computer and Communications Security, Washington, D.C., USA, October 2003.

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NETWORK SECURITY THROUGH PENETRATION TESTING

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Abstract- Each organization needs to effectively deal with major security concerns, forming a security policy according to its requirements and objectives. In anonymizing networks which provides free access to network usually hides clients IP address. Unfortunately some users have misused such networks by using it for abusive purpose. To block such a users administrators relay on router blocking or IP address blocking for disabling the misbehaving user. To recover this problem we present penetration test in which server blacklists the misbehaving user for particular time period using clients MAC address. This proactive approach is usually interpreted wrongly in only up-to-date software and hardware. Regular updates are necessary, although, not enough, because potential mis-configurations and design flaws cannot be located and patched, making the whole network vulnerable to attackers. In this paper we present how a comprehensive security level can be reached through extensive Penetration Tests.

Keywords- *Penetration testing, Network Security, Nymble*

I. INTRODUCTION

Every modern issue has to effectively deal with the security issues that arise from the technologies. An organization that truly wants to adapt a proactive approach, aggressively seeks out all types of vulnerabilities by using relevant methods with the actual hackers. This process of systematically and actively testing a deployed network to determine potential vulnerabilities is called penetration testing.

The penetration testing is divided into four phases: planning, discovery, exploitation and reporting.

- Planning phase- Initially the scope for the assignment is defined. The flow of the test is defined. After the management consent, the penetration testing team gathers crucial input about the organization operational procedures and security policies, towards defining the scope for the test.
- Discovery phase- It is also known as information gathering phase. During the information gathering process the penetration testing team launches scanning and enumeration procedures to gain as much information as possible about the target network and the participating systems and services. The gathering phase can be further divided into non-intrusive (public repositories, documents, mailing lists, web profiles etc) and intrusive (port scanning, firewall rules, matching OS fingerprints etc) inspection processes. Having adequate amount of information the testing team can profile the target network and enumerate possible exploitable vulnerabilities.
- Exploitation phase- Using as input the discovered vulnerabilities arriving from the previous phase, the penetration testing team revises matching proof-of-concept exploits that may lead to a network or service security bridge.While exploiting network vulnerabilities and mis-configurations, the testing team might discover additional information that can feedback the discovery phase. This interaction between the discovery and exploitation phases is continuous throughout the actual test.
- Reporting phase- The report writing can begin in parallel to the other three stages, although must finish after exploitation phase has been completed. A successful report details all the findings and their impacts to the organization by taking into account both the technical and management aspects in its format. It is very important to conduct a fully detailed and well documented report in order to inform the management about the security risks and provide technical details.

In earlier system instead of blocking the particular client for misbehavior the whole network associated with it is blocked. In our system we are blocking the particular client instead of blocking the whole network attached to it.

II. SYSTEM ARCHITECTURE

2.1 The Pseudonym Manager

The user must first contact the Pseudonym Manager (PM) and demonstrate control over a resource; for IP-address blocking, the user must connect to the PM directly. We assume the PM has knowledge about Tor routers, for example, and can ensure that users are communicating with it directly. Pseudonyms are deterministically chosen based on the controlled resource, ensuring that the same pseudonym is always issued for the same resource. As we will explain, the user contacts the PM only once per likability window (e.g., once a day).

2.2 The Nymble Manager

After obtaining a pseudonym from the PM, the user connects to the Nymble Manager (NM) through the anonymizing network, and requests nymbles for access to a particular server (such as Wikipedia). User's requests to the NM are therefore pseudonymous, and nymbles are generated using the user's pseudonym and the server's identity. These nymbles are thus specific to a particular user-server pair. To provide the requisite cryptographic protection and security properties, the NM encapsulates nymbles within nymble tickets.

2.3 Blacklisting a user

If a user misbehaves, the server may link any future connection from this user within the current linkability window (e.g., the same day). A user connects and misbehaves at a server during time period within linkability window. The server later detects this misbehavior and complains to the NM in time period.

Therefore, once the server has complained about a user, that user is blacklisted for the rest of the day. Even though misbehaving users can be blocked from making connections in the future, the users past connections remain unlinkable, thus providing backward unlinkability and subjective blacklisting.

2.4 Notifying the user of blacklist status

Users who make use of anonymizing networks expect their connections to be anonymous. If a server obtains a seed for that user, however, it can link those users' subsequent connections. The linkability window allows for dynamism since resources such as IP addresses can get reassigned and it is undesirable to blacklist such resources indefinitely and it ensures forgiveness of misbehavior after a certain period of time.

2.5 Blacklisting a user

If a user misbehaves, the server may link any future connection from this user within the current linkability window (e.g., the same day). Consider as an example: A userconnects and misbehaves at a server during time period.

The server later detects this misbehavior and complains to the NM in time period. As part of the complaint, the server presents the nymble ticket of the misbehaving user and obtains the corresponding seed from the NM. Even though misbehaving users can be blocked from making connections in the future, the users past connections remain unlinkable, thus providing backward unlinkability and subjective blacklisting.

2.6 Notifying the user of blacklist status

Users who make use of anonymizing networks expect their connections to be anonymous. If a server obtains a seed for that user, however, it can link that users subsequent connections. The users be notified of their blacklist status before they present a nymble ticket to a server. In our system, the user can download the servers blacklist and verify her status. If blacklisted, the user disconnects immediately. Since the blacklist is signed by the NM, the authenticity of the blacklist is easily verified if the blacklist was updated. If the user is guaranteed that he or she will not be linked if the user verifies the integrity and freshness of the blacklist before sending his or her nymble ticket.

III. LIMITATIONS

Need of refreshing

We need to refresh the database of abusive words in timely manner as the new words are entering system.

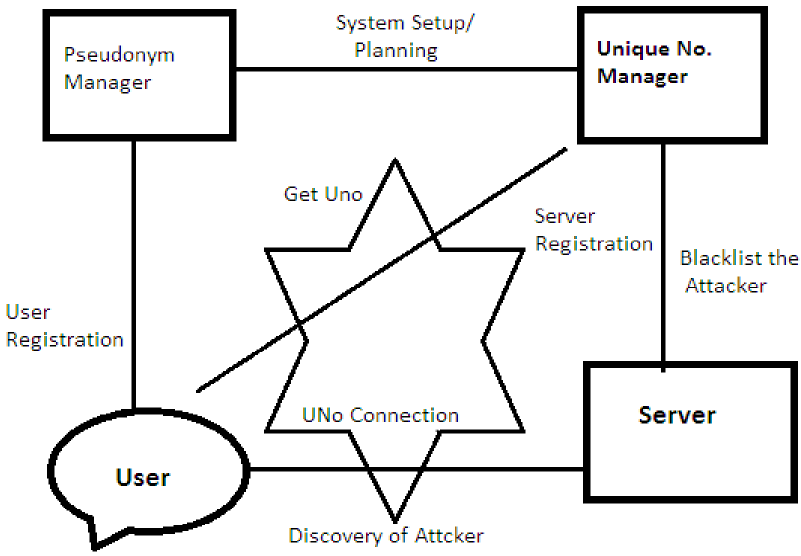

Fig 1: System Architecture

ACKNOWLEDGEMENTS

First and foremost, I would like to thank my guide, Prof. Miss Ketaki Naik, for her guidance and support.

I will forever remain grateful for the constant support and guidance extended by guide, in making this report. Through our many discussions, she helped me to form and solidify ideas.

The invaluable discussions I had with her, the penetrating questions she has put to me and the constant motivation, has all led to the development of this project.

I wish to express my sincere thanks to the Head of department, Prof. D.A.Godse also grateful thanks to Prof. Miss Ketaki Naik, Prof. S.M.Murge and the departmental staff members for their support.

I would also like to thank to my friends for listening to my ideas, asking questions and providing feedback and suggestions for improving my ideas.

REFERENCES

[1] Test Input Generation for Java Containers Using State Matching; W. Visser, C.S.Pasareanu, and R. Pelanek in Proc. Intl Symp. on Software Testing and Analysis, 2006, pages 37-48.

[2] P.P. Tsang, M.H. Au, A.Kapadia, and S.W. Smith, "Black-listable Anonymous credentials: Blocking Misbehaving Users without TTPs", proc. 14th ACM conf. Computer and Comm. Security(CCS'07), pp.72-81,2007.

[3] P.C. Johnson, A.Kapadia, P.P. Tsang, and S.W. Smith, "Nymble: Anonymous IP-Address Blocking", Proc. Conf.. Privacy Enhancing Technologies, Springer, pp. 113-133,2007.

[4] J.E. Holt and K.E. Seamsons, "Nym: Practical Pseudonymity for Anonymous Networks", Internet Security Research Lab Technical Report 2006-4, BrighamYoung Univ., June 2006.

[5] C. Cornelius, A. Kapadia, P.P Tsang, and S.W. Smith, "Nymble: Blocking Misbehaving Users in Anonymizing Networks", Technical Report TR2008-637, Dartmouth College, Computer science, Dec. 2008.

[6] Bishop, M., "About Penetration testing",IEEE Security and Privacy(2007),Volume: 5,Issue: 6.

[7] Hamisi, N.Y.,Mvungi, N.H., Mfinanga,D.A. and Mwinyiwiwa,B.M.M.,"Intrusion detection by penetration test in an organization network",ICAST 2009.

INVESTIGATION OF TCP CONGESTION WINDOW IN ADHOC NETWORK

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Abstract- In MANET nodes have forward each other packet through the network without any fixed communication infrastructure. A contention problem occurs in network when adjacent nodes shared channel to transmit packets, Medium contentions cause network congestion because of a lack of coordination between the transport layer and the medium access layer. Due to the channel interference the bandwidth delay produces reach to its maximum value. Packet loss occurs at MAC layer due to congestion. Many approaches have been proposed to overcome these difficulties. This chapter is a survey on TCP performance in mobile ad-hoc networks. We describe the problems of standard TCP in ad-hoc networks, and then present the design space and existing solutions to improve TCP throughput.

Keywords- *Congestion Control, Contention detection, Bandwidth delay, Mobile Adhoc network, channel utilization. Introduction.*

I. INTRODUCTION

Each node in a MANET is capable of moving independently and functioning as a router that discovers and maintains routes and forwards packets to other nodes. Thus, MANETs are multi-hop wireless networks [1] by nature. Transmission control protocol (TCP) is a transport layer protocol [2] which provides reliable end-to- end data delivery between end hosts in traditional wired network environment. In TCP, reliability is achieved by retransmitting lost packets. Thus, each TCP sender maintains a running average of the estimated round trip delay and the average deviation derived from it. Packets will be retransmitted if the sender receives no acknowledgment within a certain timeout interval (e.g., the sum of smoothed round trip delay and four times the average deviation) or receives duplicate acknowledgments. Due to the inherent reliability of wired networks, there is an implicit assumption made by TCP that any packet loss is due to congestion. To reduce congestion, TCP will invoke its congestion control[3,4] mechanisms whenever any packet loss is detected. Since TCP is well tuned, it has become the de facto transport protocol in the Internet that supports many applications such as web access, file transfer and email. Due to its wide use in the Internet, it is desirable that TCP remains in use to provide reliable data transfer services [5] for communications within wireless networks and for those across wireless networks and the wired Internet. It is thus crucial that TCP performs well over all kinds of wireless networks in order for the wired Internet to extend to the wireless world. To understand TCP behavior and improve TCP performance over wireless networks, given these wireless specific challenges, considerable research has been carried out and many schemes have been proposed. As the research in this area is still active and many problems are still wide open, this serves to pinpoint the primary causes for TCP performance degradation over wireless networks, and cover the state of the art in the solution spectrum, in hopes that readers can better understand the problems and hence propose better solutions based on the current ones.

The basic functions of TCP as a transport layer protocol include flow control, error recovery and congestion control[5,6], while the state-of-the-art techniques include fast retransmission and recovery, selective acknowledgment, etc., mainly focusing on how to promptly and effectively respond to network congestion.

By examining the TCP's performance studies over MANETs we identified the following major problems:

- TCP is unable to distinguish between losses due to route failures and network congestion.
- TCP suffers from frequent route failures.

A. Challenges in TCP communication

Wireless ad hoc networks have some inherent adverse characteristics [7] that will significantly deteriorate TCP performance if no action is taken. In essence, these characteristics include bursty channels errors, mobility and communication asymmetry.

1) Channel Errors

In wireless channels, relatively high bit error rate because of multipath fading and shadowing may corrupt packets in transmission, leading to the losses of TCP data segments or ACKs. If it cannot receive the ACK within the retransmission timeout, the TCP sender immediately reduces its congestion window to one segment, exponentially backs off its RTO and retransmits the lost packets. Intermittent channel errors may thus cause the congestion window size at the sender to remain small, thereby resulting in low TCP throughput.

2) Mobility

Cellular networks are characterized by handoffs due to user mobility. Normally, handoffs may cause temporary disconnections, resulting in packet losses and delay. TCP will suffer a lot if it treats such losses as congestion and invokes unnecessary congestion control mechanisms. Those handoffs are expected to be more frequent in next generation cellular networks as the micro cellular structure is adopted to accommodate an increasing number of users. Thing could be worse if TCP cannot handle handoffs gracefully. Similar problems may occur in wireless LAN, as mobile users will also encounter communication interruptions if they move to the edge of the transmission range of the access point.

3) Medium contention, Hidden terminal, and Exposed terminal problem*s*

Contention-based medium access control (MAC) schemes, such as IEEE 802.11 MAC protocol, have been widely studied and incorporated into many wireless test beds and simulation packages for wireless multi-hop ad hoc networks, where the neighboring nodes contend for the shared wireless channel before transmitting. There are three main problems, i.e., the hidden terminal problem, the exposed terminal problem, and unfairness. A hidden node is the one that is within the interfering range of the intended receiver but out of the sensing range of the transmitter. The receiver may not correctly receive the intended packet due to collision from the hidden node. An exposed node is the one that is within the sensing range of the transmitter but out of the interfering range of the receiver. Though its transmission does not interfere with the receiver, it could not start transmission because it senses a busy medium, which introduces spatial reuse deficiency. The binary exponential backoff scheme always favors the latest successful transmitter, and hence results in unfairness. These problems could be more harmful in multi-hop ad hoc networks than in Wireless LANs as ad hoc networks are characterized by multi-hop connectivity.

II. PROPOSALS TO IMPROVE TCP PERFORMANCE IN AD HOC NETWORK

In this section we present the various proposals which have been made in the literature to improve the performance of TCP in Ad hoc networks.Adel Gaafar A.Elrahim et.al [8] describes a modification to the TCP congestion control for use in wireless sensor networks. It shows that by slightly modifying the algorithm of the TCP, it can be made to respond better to wireless links. It presents enhancement method by increasing RTO value. A TCP sender constantly tracks the Round Trip Time (RTT) for its packets and uses a timeout mechanism to trigger retransmissions if an ACK is not received before the timer expires. As a de facto standard, TCP sender uses the tracked average (RTT) plus m times the mean deviation of RTTs as the RTO value [7] for the next packet where the typical value of the factor m is 4. More precisely let RTT (k) denote the k-th measurement value of RTT. Where its value is the time interval between the beginnings of the packet transmission until all ACK for the packet is received by the sender. Yao-Nan Lien et al [9] proposed a new TCP congestion control mechanism by router-assisted approach. Their proposed TCP protocol, called TCP Muzha uses the assistance provided by routers to achieve better congestion control. To use TCP Muzha, routers are required to provide some information allowing the sender to estimate more accurately the remaining capacity over the bottleneck node with respect to the path from the sender to the receiver. With this information, TCP Muzha will be able to enhance the performance of both TCP and network.

Wei Sun et al [10] have compared the general AIMD-based congestion control mechanism (GAIMD) with Equation-based congestion control mechanism (TFRC TCP-Friendly Rate Control) over a wide range of MANET scenario, in terms of throughput fairness and smoothness. Their results have shown that TFRC and GAIMD are able to maintain throughput smoothness in MANET, but at the same time, they require only a less throughput than the competing TCP flows. Also their results show that TFRC changes its sending rate more smoothly than GAIMD does, but it gets the least throughput compares with TCP and GAIMD.
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Yung Yi et al [11] have developed a fair hop-by-hop congestion control algorithm with the MAC restriction being imposed in the form of a channel access time constraint, using an optimization based structure. In the lack of delay, they have shown that their algorithm is globally stable using a Lyapunov-function-based approach. Next, in the presence of delay, they have shown that the hop-by-hop control algorithm has the property of spatial spreading. Also they have derived bounds on the “peak load” at a node, both with hop-by-hop control, as well as with end-to-end control, show that significant gains are to be had with the hop-by-hop scheme, and validate the analytical results with simulation.

Umut Akyol et al [12] have studied the problem of jointly performing scheduling and congestion control in mobile ad hoc networks so that network queues remain bounded and the resulting flow rates satisfy an associated network utility maximization problem.

They have defined a specific network utility maximization problem which is appropriate for mobile ad hoc networks. They have described a wireless Greedy Primal Dual (wGPD) algorithm for combined congestion control and scheduling that aims to solve this problem. They have shown how the wGPD algorithm and its associated signaling can be implemented in practice with minimal disruption to existing wireless protocols. In greedy algorithm the chances of best solution are minimum and also not sure about the solution is best.

S.Karunakaran et al [13] have presented a Cluster Based Congestion Control (CBCC) protocol that consists of scalable and distributed cluster-based mechanisms for supporting congestion control in mobile ad hoc networks. The distinctive feature of their approach is that it is based on the self-organization of the network into clusters. The clusters autonomously and proactively monitor congestion within its localized scope.

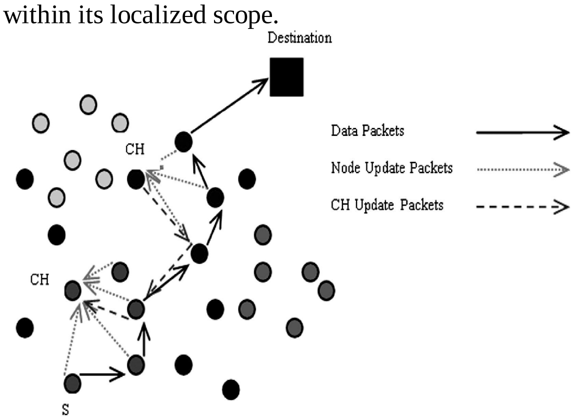

Figure 1 An Overview of CBCC Network Structure.

S.Venkatasubramanian et al [14] proposed the QoS architecture for Bandwidth Management and Rate Control in MANETs. The bandwidth information in the architecture can be used for QoS capable routing protocols to provide support to admission control. The traffic is balanced and the network capacity is improved as the weight value assists the routing protocol to evade routing traffic through congested area. The source nodes then perform call admission control for different priority of flows based on the bandwidth information provided by the QoS routing. In addition to this, a rate control mechanism is used to regulate best-effort traffic, whenever network congestion is detected. In this mechanism, the packet generation rate of the low-priority traffic is adjusted to incorporate the high-priority traffic.

R.Mynuddin Sulthani et al [15] proposed a joint design of reliable QoS architecture for mobile ad hoc networks. In the reliable multipath routing protocol, dispersion and erasure code techniques are utilized for producing replicated fragments for each packet, to enhance reliability. Then messages with good delivery probability are identified and transmitted through the paths with high average node delivery index. While it receives an assured number of fragments, destination can recover the original packet. Next, a call admission control (CAC) scheme has been developed, in which, the calls are admitted based on the bandwidth availability of the path. Once congestion occurs, the best effort traffic is rate controlled, to free bandwidth for the real-time flows. In CAC the overhead is to generate congestion alert packets in just before its probability of congestion

Lijun Chen et al [16] proposed the joint design of congestion control, routing and scheduling for ad hoc wireless networks. They formulate resource allocation in the network with fixed wireless channels or single-rate wireless devices as a utility maximization problem with schedulability and rate constraints arising from contention for the wireless channel.

We also extend the dual algorithm to handle the network with time-varying channel and adaptive multi-rate devices, and surprisingly show that, despite stochastic channel variation, it solves an ideal reference system problem which has the best feasible rate region at link layer. In future, they will extend the results to networks with more general interference models and/or node mobility and further will enhance the performance gain from cross-layer design involving link layer.

Xuyang Wang et al [17] proposed a cross layer hop by hop congestion control scheme to improve TCP performance in multihop wireless networks which coordinates the congestion response across the transport, network, and transport layer protocols. The proposed scheme attempts to determine the actual cause of a packet loss and then coordinates the appropriate congestion control response among the MAC network, and transport protocols. The congestion control efforts are invoke at all intermediate and source node along the upstream paths directed from the wireless link experiencing the congestion induced packet drop.

Kazuya Nishimura et al [18] proposed a routing protocol that reduces network congestion for MANET using multi-agents. They use two kinds of agents: Routing Agents to collect information about congestion and to update the routing table at each node, and Message Agents to move using this information. In the future, they will explore a better estimation function and discuss the limits of its effectiveness. The evaluation function itself may change depending on the environment. Incorporating learning into the function is also an interesting issue.

Bhadauria, Sharma et al [19] proposed the information about network congestion is collected and distributed by mobile agents (MA). The MA measures the queue length of the various traffic classes and the channel contention and estimates the total congestion metric to find the minimum congestion level in the network. The congestion metric is applied in the routing protocol to select the minimum congested route.

Vinay Rishiwal et al [20] proposed QoS based power aware routing protocol (Q-PAR). The selected route is energy stable and satisfies the bandwidth constraint of the application. The protocol Q-PAR is divided in to two phases. In the first route discovery phase, the bandwidth and energy constraints are built in into the DSR route discovery mechanism. In the event of an impending link failure, the second phase, a repair mechanism is invoked to search for energy stable alternate path locally. Moreover the local repair mechanism was able to find an alternate path in most of the cases enhanced the network lifetime and delayed the repair and reconstruction of the route.

III. CONCLUSION AND FUTUREWORK

As the assumption made by TCP that any packet loss is due to network congestion is no longer valid in wireless networks, TCP performs poorly in such networks. In this chapter, we point out the major reasons for this performance degradation. In particular, factors such as mobility, wireless channels and handoffs result in the poor TCP performance over single and multi hop ad hoc networks, while, aside from these factors, other factors such as medium access contention, frequent route changes, and breakages are considered to lead to the poor TCP performance in networks. Although some encouraging improvements have been reported by employing the proposed schemes, none of them can work well in all scenarios and meet all the challenges mentioned. Therefore, there is still much work to be done in the near future. To provide as supervision for future research, some critical issues regarding improving TCP performance and fairness are identified.

REFERENCES

[1] A. I. Niq and Barakatc, "Enhancing IEEE 802. 11 MAC in congested environments," in ASWN, 2004.

[2] Yuh-Shyan Chen, Athanasios V. Vasilakos and Chien-Chung Shen: Guest Editorial-Cross-Layer Design for Future Generation Wireless Networks, Wireless Pers Commun (2009) 51:375–378, DOI 10.1007/s11277-009-9759-3, published online: 26 June 2009. © Springer Science+Business Media, LLC. 2009.

[3] V. Jacobson, "Modified TCP Congestion Avoidance Algorithm," Technical report, Apr. 1990.

[4] W. Stevens, "TCP Slow Start, Congestion Avoidance, Fast Retransmit, and Fast Recovery Algorithms," *IETF RFC 2001*, 1997.

[5] S. Ryu, C. Rump, and C. Qiao, "Advances in Internet congestion control," IEEE Communications Surveys and Tutorials, vol. 3, pp. 28–39, 2003.

[6] S. Choi, K. Park, and C. Kim, "Performance Impact of Inter-Layer Dependence in Infrastructure WLANs," IEEE Trans. Mobile Computing, vol. 5, no. 7, pp. 829-845, 2006.

[7] W. Richard Stevens, TCP/IP Illustrated, Volume 1. Protocol Addison-Wesley, New York, Jan- 1994

[8] Adel Gaafar A.Elrahim, Hussein A.Elsayed, Salwa El Raml, Magdy M, Ibrahim "Improving TCP Congestion Control for Wireless Sensor Networks" 978-1-4577-0972-2/11/2011 IEEE.

[9] Lien, Y.N., Hsiao, H.C.: A New TCP Congestion Control Mechanism over Wireless Ad Hoc Networks by Router-Assisted Approach. 27th IEEE International Conference on Distributed Computing Systems Workshops. (2007).

[10] Sun, W., Wen, T., Guo, Q.: A Performance Comparison of Equation-Based and GAIMD Congestion Control in Mobile Ad Hoc Networks. International Conference on Computer Science and Software Engineering. (2008).

[11] Yi, Y., Shakkottai, S.: Hop-by-Hop Congestion Control Over a Wireless Multi-Hop Network. IEEE/ACM Transactions on Networking. (2007)

[12] Akyol, U., Andrews, M., Gupta, P., Hobby, J., Saniee, I., Stolyar, A.: Joint Scheduling and Congestion Control in Mobile Ad-Hoc Networks. Proceedings of IEEE INFOCOM. (2008).

[13] Karunakaran, S., Thangaraj, P.: A Cluster Based Congestion Control Protocol for Mobile Adhoc Networks. International Journal of Information Technology and Knowledge Management. Vol. 2, No. 2, pp. 471-474.(2010).

[14] Venkatasubramanian, S., Gopalan, N.P.: A Quality of service architecture for resource provisioning and rate control in mobile ad hoc networks. International Journal of Ad hoc, Sensor & Ubiquitous Computing (IJASUC). Vol.1, No.3.(2010).

[15] Sulthani, R.M., Rao, D.S.: Design of an Efficient QoS Architecture (DEQA) for Mobile Ad hoc Networks. ICGST-CNIR Journal. Vol. 8, Issue 2. (2009).

[16] Chen, L., Lowy, S.H., Chiangz, M., Doyley, J.C.: Cross-layer Congestion Control, Routing and Scheduling Design in Ad Hoc Wireless Networks. Proc., IEEE, 25th international conference on computer communication, INFOCOM. pp 1 – 13. (2007).

[17] Wang, X., Perkins, D.: Cross-layer Hop-by-hop Congestion Control in Mobile Ad Hoc Networks. Proc., IEEE, wireless communication and networking conference, WCNC. pp 2456 – 2461. (2008).

[18] Nishimura, K., Takahashi, K.: A Multi-Agent Routing Protocol with Congestion Control forMANET. Proceedings 21st European Conference on Modeling and Simulation (ECMS). (2007).

[19] Bhadauria SS,Sharma V., Framework and Implementation of an Agent Based Congestion ControlTechnique for Mobile Ad-hoc Network ,ICAC3 2011, CCIS 125, pp. 318–327, 2011.

[20] Vinay Rishiwal, Shekhar Verma and S. K. Bajpai, "QoS Based Power Aware Routing in MANETs," International Journal of Computer Theory and Engineering (IJCTE), vol. 1, no. 1, pp. 47-54, 2009.

LOSSLESS MEDICAL IMAGE COMPRESSION

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Abstract- Modern medical imaging requires storage of large quantities of digitized clinical data. Due to the constrained bandwidth and storage capacity, however, a medical image must be compressed before transmission and storage. Context-based Adaptive Lossless Image Codec (CALIC) is one of the most efficient lossless encoding techniques for continuous-tone images.. CALIC is an efficient context-based lossless image encoding technique. For each context, an error feedback operation is performed by calculating a bias value. This is added to the value that is initially predicted by a less sophisticated predictor.

Keywords- lossless image compression, medical images, high bit depth images, Medical Imaging, CALIC.

I. INTRODUCTION

Data compression is the technique to reduce the redundancies in data representation in order to decrease data storage requirements and hence communication costs. Reducing the storage requirement is equivalent to increasing the capacity of the storage medium and hence. communication bandwidth. Thus the development of efficient compression techniques will continue to be a design challenge for future communication systems and advanced multimedia applications. Data is represented as a combination of information and redundancy.(1) Information is the portion of data that must be preserved permanently in its original form in order to correctly interpret the meaning or purpose of the data. Redundancy is that portion of data that can be removed when it is not needed or can be reinserted to interpret the data when needed. Most often, the redundancy is reinserted in order to generate the original data in its original form. A technique to reduce the redundancy of data is defined as Data compression. The redundancy in data representation is reduced such a way that it can be subsequently reinserted to recover the original data, which is called decompression of the data. There are two major approaches in the image compression field, namely: lossless and lossy.

II. LOSSLESS COMPRESSION

The Lossless Data Compression technique recommended preserves the source data accuracy by removing redundancy from the application source data. In the decompression processes the original source data is reconstructed from the compressed data by restoring the removed redundancy.(3)

The reconstructed data is an exact replica of the original source data. The quantity of redundancy removed from the source data is variable and is - highly dependent on the source data statistics, which are often non-stationary.In lossless compression, compressing and decompressing an image result in an exact replica of the original image. (4)There are various applications where any loss in image information is unacceptable, e.g. medical imaging. In lossy compression, some image information is discarded in order to achieve better compression. This only allows a close replica of the original image to be reconstructed from the compressed data. Lossless compression is considered the base from which lossy compression algorithms are derived by means of a suitable quantization scheme. Hence, improvement in lossless compression schemes should also benefit lossy compression ones. The compression scheme presented in this paper is a lossless.

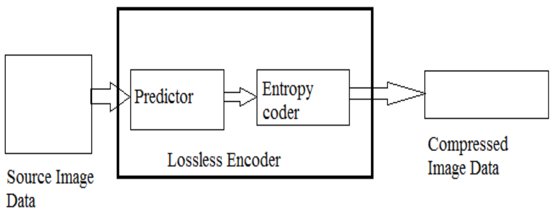

Fig.1 Lossless Image Compression

Predictive encoding is a major class of encoding schemes that is utilized in lossless compression. Compression is accomplished by making use of the previously encoded pixels that are available to both the encoder and the decoder in order to predict the value for the next pixel to be encoded. Instead of the actual pixel value, the prediction error is then encoded.(11) Context-based prediction is a kind of adaptive predictive encoding in which pixels are classified into different classes (a.k.a. contexts) based on pixel neighbourhood characteristics. A suitable predictor for each context is adaptively selected and utilized for each context.

The Need for Compression- With the advance development in Internet and multimedia technologies, the amount of information that is handled by computers has grown exponentially over the past decades. This information requires large amount of storage space and transmission bandwidth that the current technology is unable to handle technically and economically. One of the possible solutions to this problem is to compress the information so that the storage space and transmission time can be reduced.

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III. CALIC

CALIC is a single pass context-based adaptive predictor scheme. The operation of CALIC can be divided into two modes, namely: binary mode and continuous-tone mode. The selection between these two modes is performed on the fly based on context.(2) Prediction that utilizes priorities knowledge of image smoothness. The GAP is simple, adaptive, nonlinear predictor, which can adapt itself to the intensity gradients near the predicted pixel; it weights the neighboring pixels of current sample according to the estimated gradients of the image Let us denote value of current pixel as I[i, j].

In context classification, each pixel is classified to one of the 576 predefined contexts. The context selection is based on comparing the value of the initial prediction with the pixel neighborhood's values. For each context, CALIC assumes that the GAP predictor is consistently repeating a similar prediction error.(10) To compensate for this error, CALIC incorporates an error feedback stage, at which a bias value is added to the initial prediction. This bias value is the expectation of the prediction errors at the pixel's context. Prediction errors are entropy encoded using arithmetic encoder that utilizes context conditional probabilities, i.e., P(error|context).

IV. COMPRESSION EFFICIENCY

Compression efficiency is measured for lossless and lossy compression. For lossless coding it is simply measured by the achieved compression ratio for each one of the test images.

V. RESULTS

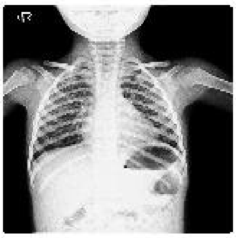
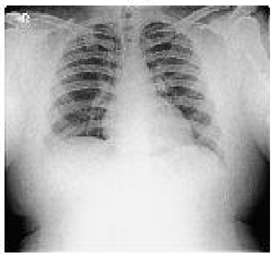
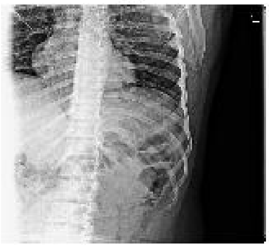
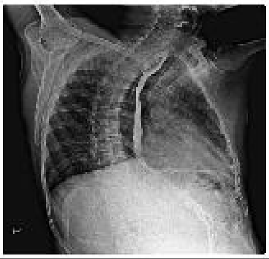
Images	Compression ratio
Image:1 	C. R. : 0.364614
Image:2 	C.R.: 0.255417
Image:3 	C.R.: 0.226842
Image:4 	C.R.: 0.444703
Image:5 	C.R.: 0.543056
Image:6 	C.R.: 0.475178
Image:7 	C.R.: 0.351681

Image:8 	C.R.: 0.395555
Image:9 	C.R.: 0.431548
Image:10 	C.R.: 0.431548
Image:11 	C.R.: 0.365668
Image:12 	C.R.: 0.324280

VI. CONCLUSION

In this paper we propose a lossless prediction scheme of Context Based Adaptive Lossless Image Coding (CALIC). Context-based Adaptive Lossless Image Codec (CALIC) is one of the most efficient lossless encoding techniques for continuous-tone images. However, its performance is considerably downgraded on images with fewer and widely separated grey levels. Predictive encoding is a major class of encoding schemes that is utilized in lossless compression.

REFERNCES

[1] D. A. Clunie, “Lossless compression of grayscale medical images - effectiveness of traditional and state of the art approaches”, Proc. SPIE, Medical Imaging 2000, 3980:74-84, 2000.

[2] Xiaolin Wu, Nasir Memon, “CALIC- A context based lossless image codec”, Department of computer science, Canada

[3] ISO/IEC, “Lossless and near-lossless compression of continuous-tone still images (JPEG-LS)”, ISO/IEC FCD 14495-1 – public draft, 1997.

[4] P.G. Howard and J.S. Vitter, “Fast and efficient lossless image compression” In J.A. Storer and M. Cohn, editors, proceedings of the Data Compression Conference, pages 351-360. IEEE computer society pres,1993

[4] G.G. Langdon. Sunset: a hardware oriented algorithm for lossless compression of gray scale images. In medical Imaging V: Image capture, formatting and Display, Volume 1444, pages 272-282. SPIE, 1991

[5] N.D. Memon and K. Sayood, “Lossless image compression-a comparative study. In still image compression” pages 8-20. SPIE Proceedings volume 2218,1995

[6] X. Wu, Context-Based, Adaptive, “Lossless Image Coding”, IEEE Trans. Communications, vol. 45, No. 4, April 1997

[7] X. Wu, Lossless compression of continuous-tone images via context selection, quantization, and modeling, IEEE Trans. Image Processing, vol. 6, May 1997.

[8] A. Moffat, R. Neal, and I. Witten, “Arithmetic coding revisited”, Data Compression Conf., DCC '95. Proceedings, 28-30 March 1995, pp. 202-211.

[9] S. Todd, G. G. Langdon, and J. J. Rissanen, “Parameter reduction and context selection for compression of gray scale images”, IBM J. Res. Develop., vol. 29, no. 2, pp. 188-193, 1985.

[10] Aleksej Avramović, Branimir Reljin, “Gradient Edge Detection Predictor for Image Lossless Compression”, 52nd International Symposium ELMAR-2010, 15-17 September 2010, Zadar, Croatia

[11] Farshid Sepehrband, Mohammad Mortazavi, Seyed Ghorashi, “An efficient lossless medical image transformation method by improving prediction model”, 978-1-4244-5900-1/10/$26.00 ©2010 IEEE.

COSINE SIMILARITY – A PROMISING CRITERIA FOR EXTRACTING MAIN CONTENT FROM WEB DOCUMENTS USING CONTENT STRUCTURE TREE: A SURVEY

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Abstract- Because of the use of growing information, web mining has become a primary necessity of world. Due to this, research on web mining has received a lot of interest from both industry and academia. Mining and prediction of user's web browsing behaviors and deducing the actual content in a web document is one of the active subjects. The information on web is dirty. Apart from useful information, it contains unwanted information such as copyright notices and navigation bars that are not part of main contents of web pages. These seriously harm Web Data Mining and hence, need to be eliminated. This survey aims at studying the possible similarity criteria based on cosine similarity to deduce which parts of content are more important than others. In this paper, an algorithm is proposed that extracts the main content from the web documents. The algorithm is based on Content Structure Tree (CST). Firstly, the proposed system uses HTML Parser to construct DOM (Document Object Model) tree from which it constructs Content Structure Tree (CST) which can easily separate the main content blocks from the other blocks. The proposed system can rank pages on the basis of similarity of pages to particular query.

Keywords*- Cosine Similarity; CST tree; DOM tree; Web Content Mining.*

I. INTRODUCTION

Information is expanding rapidly and hence, the web. Web is a collection of abundant information. The information on web is dirty. Apart from useful information, it contains unwanted information such as copyright notices and navigation bars that are not part of main contents of web pages. Although these information item are useful for human viewers and necessary for the Web site owners, they can seriously harm automated information collection and Web data mining, e.g. Web page clustering, Web page classification, and information retrieval. So how to extract the main content blocks become very important. Web pages contain Div block, Table block or other HTML blocks. This paper aims at extracting the main content from a web document which is relevant to user's query. Here, a new algorithm is proposed to extract the informative block from web page based on DOM-based analysis and Content Structure Tree (CST). Then we apply cosine similarity measure to identify and separate the rank of the each block from the page. Further, we also use TF-IDF (Term Frequency and Inverse Document Frequency) scheme for calculating the weight of each node on the CST. Finally, we extract the most relevant information to the query.We will discuss two algorithms in this paper. One is for generating Content Structure Tree and another for extracting main content from Content structure Tree.

II. LITERATURE SURVEY

From time to time, many extraction systems have been developed. In [1], Swe Swe Nyein proposed the-method of mining contents in web page using cosine similarity. In [2], C. Li et al. propose a method to extract informative block from a web page based on the analysis of both the layouts and the semantic information of the web pages. They needed to identify blocks occurring in a web collection based on the Vision-based Page Segmentation algorithm. In [3], L. Yi et al. propose a new tree structure, called Style Tree to capture the actual contents and the common layouts (or presentation styles) of the Web pages in a Web site. Their method can difficult to capture the common presentation for many web pages from different web sites. In [4], Y. Fu et al. propose a method to discover informative content block based on DOM tree. They removed clutters using XPath. They could remove only the web pages with similar layout. In [5], P. S. Hiremath et al. propose an algorithm called VSAP (Visual Structure based Analysis of web Pages) to exact the data region based on the visual clue (location of data region / data records / data items / on the screen at which tag are rendered) information of web pages. In [6] S. H. Lin et al. propose a system, InfoDiscoverer to discover informative content blocks from web documents. It first partitions a web page into several content blocks according to HTML tag <TABLE>. In [7] D. Cai et al. propose a Vision-based Page Segmentation (VIPS) algorithm that segments web pages using DOM tree with a combination of human visual cues, including tag cue, color cue, size cue, and others. In [8], P. M. Joshi propose an approach of combination of HTML DOM analysis and Natural Language Processing (NLP) techniques for automated extractions of main article with associated images form web pages. Their approach did not require prior knowledge of website templates and also extracted not only the text but also associated images based on semantic similarity of image captions to the main text. In [9], Y. Li et al. propose a tree called content structure tree which captured the importance of the blocks. In [10], R, R, Mehta propose a page segmentation algorithm which used both visual and content information to obtain semantically meaningful blocks. The output of the algorithm was a semantic structure tree. In [11], S. Gupta proposes content extraction technique that could remove clutter without destroying webpage layout. It is not only extract information from large logical units but also manipulate smaller units such as specific links within the structure of the DOM tree. Most of the existing approaches based on only DOM tree.

III. WEB MINING

Web Mining is the extraction of interesting and potentially useful patterns and implicit information from artifacts or activity related to the World Wide Web. It has become a necessity as the information across the world is increasing tremendously and hence the size of the web.

According to the differences of the mining objects, there are roughly three knowledge discovery domains that pertain to web mining: Web Content Mining, Web Structure Mining and Web Usage Mining. Web usage mining is an application of data mining technology to mining the data of the web server log file. It can discover the browsing patterns of user and some kind of correlations between the web pages. Web usage mining provides the support for the web site design, providing personalization server and other business making decision, etc. Web mining applies the data mining, the artificial intelligence and the chart technology and so on to the web data and traces users' visiting characteristics, and then extracts the users' using pattern. The web usage mining generally includes the following several steps:

1. Data collection
2. Data pretreatment
3. Establishing interesting model
4. Pattern analysis

Web mining algorithm based on web usage mining is also produces the design mentality of the electronic commerce website application algorithm. This algorithm is simple, effective and easy to realize, it is suitable to the web usage mining demand of construct a low cost Business-to-Customer (B2C) website. Web mining algorithm generally includes the following several steps:

1. Collect and pre-treat users' information
2. Establish the topology structure of web site
3. Establish conjunction matrices of the users visit pages
4. Concrete application.

A. Web Content Mining: Web Content Mining refers to description and detection of useful information from the web contents / data / documents. There are two views on web content mining: view of information retrieval and data base. The aim of web content mining according to data retrieval based on content is to help the process of data filtering or finding data for the user which is usually performed based on extraction or demand of users; while according to the view of data bases it means attempt for modeling the data on web and its combination such that most of the expert query required for searching the information can be executed on this kind of data mode.

B. Web Structure Mining: Web Structure Mining tries to discover the model underlying the link structures of the web. This model can be used to categorize web pages and is useful to generate information such as the similarity and relationship between different web sites.

C. Web Usage Mining: Web Usage Mining using the data derived from using effects on the web detects the behavioral models of the users to access the web services automatically.

IV. SYSTEM FLOW

A web page usually contains main content blocks and noise content blocks. Only the main content blocks represent the informative part that is really we want to know. Thus, in the first task, the HTML Parser is used to create a DOM tree representation of the original html source. Second task is to find relevant information from the web pages in the site. So, a Content Structure Tree (CST) is created based on the DOM tree. Then cosine similarity method is used to evaluate each node in the content structure tree and it can easily get the target informative block. The system architecture is shown in Fig. 1.

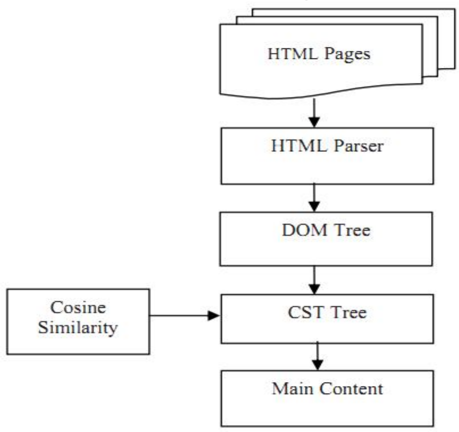

Fig. 1. System Architecture

V. NOISE ELIMINATION AND CONTENT STRUCTURE TREE

Content Structure Tree is used for Noise Elimination from the previously generated DOM (Document Object Model) tree. Content Structure Tree is an improvement over traditional approach which uses Style Tree to determine the unwanted content. Content Structure Tree deals with actual content of the web pages while the Style Tree is concerned more about the similarities in presentation styles of the documents. The traditional approach uses Clustering Method.

The popular k-means clustering algorithm is used in this technique. All the 5 categories of Web pages are put into a big set, and the clustering algorithm was used to cluster them into 5 clusters. Since the k-means algorithm selects the initial cluster seeds randomly, a large number of experiments (800) are performed to show the behaviors of k-means clustering before and after page cleaning.

Clustering results after SST based cleaning are dramatically better than the results using the original noisy Web pages. This method also helps to produce much better clustering results than the template based method. Fig. 2. gives the statistics of F scores over the 800 clustering runs using the original Web pages, the pages cleaned with the template based method and the pages cleaned with the SST based method respectively. It is observed that over the 800 runs, the average F score for the noise case (without cleaning) is 0.506, and the average F score for the template based cleaning case is 0.631, while the average F score for the SST based cleaning case is 0.751, which is a remarkable improvement.

Method	Ave(F)	F < 0.5	F >= 0.7	F >= 0.8	F >= 0.9
F(N)	0.506	47.63%	0.50%	0.13%	0.00%
F(T)	0.631	10.63%	23.25%	7.75%	0.00%
F(S)	0.751	3.25%	78.13%	24.75%	11.75%

Fig. 2. Statistics of k-means clustering results

More specifically, before cleaning, only 0.5% of the 800 results (4 out of 800) have the F scores no less than 0.7, and 47.63% lower than 0.5. After template based cleaning, 23.25% of the 800 clustering results have the F scores no less than 0.7, and 10.63% lower than 0.5. While after the SST based cleaning, 78.13% of the 800 results have F scores no less than 0.7, and only 3.25% lower than 0.5. Thus, we can conclude that our proposed noise elimination method is much more effective than the template based method for Web page clustering.

VI. COSINE SIMILARITY

Cosine similarity is a measure of similarity between two high-dimensional vectors. In essence, it is the cosine value of the angle between two vectors.

The similarity of content in the web pages is estimated using cosine similarity measure which is the cosine of the angle between the query vector q and the document vector d_j. Then the weight of each node (term) in CST tree such as Text node, Image node, and Link node is calculated. TF-IDF scheme (Term Frequency and Inverse Document Frequency) is used to calculate the weight. The weight of a term t_i in document d_j is the number of times that appears in document. In this scheme, an arbitrary normalized w_{ij} is defined. Then, the weight of content node is obtained, Content Weight = Text Weight + Image Weight + Link Weight

The content value which is computed by the children nodes of HtmlItem node will be added to element which is the first child of HtmlItem node. Before adding to the parent node, this system checks the HtmlItem nodes whether there are same level nodes. If so, then similarity of these nodes is computed using their weights and then all HtmlItem nodes are eliminated, except the highest similarity node. Ranking of the documents is done using their similarity values. The top ranked documents are regarded as more relevant to the query.

VII. CONCLUSION

Many researchers have been developed several approaches for extract main content from web pages. Most of the approaches based on the only DOM tree. This system is based on the CST tree generated by the DOM tree and also could extract the relevant documents from the web pages using cosine similarity measure. A commercial Web page typically contained many information blocks. Apart from the main content blocks, it usually has such blocks as navigation panels, copyright and privacy notices, and advertisements which are called noisy blocks. These noisy blocks can seriously harm Web data mining. This approach can be extended to detection and removal of noises on web pages and also can be used to extract the main content.

REFERENCES

[1] Swe Swe Nyein, “Mining Contents in Web Page Using Cosine Similarity”, IEEE, Copyright 2011.

[2] C. Li, J. Dong, and J. Chen, “Extraction of Informative Blocks from Web Pages Based on VIPS”, 1553-9105/ Copyright January 2010.

[3] L. Yi, B. Liu, and X. Li, “Eliminating Noisy Information in Web Pages for Data Mining”, in Proc. ACM SIGKDD International Conference on Knowledge Discovery & Data Mining (KDD-2003).

[4] Y. Fu, D. Yang, and S. Tang,”Using XPath to Discover Informative Content Blocks of Web Pages”, IEEE. DOI 10.1109/SKG, 2007.

[5] P. S. Hiremath, S. S. Benchalli, S. P. Algur, and R. V. Udapudi, “Mining Data Regions from Web Pages”,

International Conference on Management of Data COMAD, India, December 2005.

[6] S. H. Lin and J. M .Ho, "Discovering Informative Content Blocksfrom Web Documents", in Pro. ACM SIGKDD International Conference on Knowledge Discovery & Data Mining, pp.588-593, July 2002.

[7] D. Cai, S. Yu, J. R. Wen, and W. Y. Ma, "VIPS: a Vision-based PageSegmentation Algorithm", Technical Report, MSR-TR, Nov. 1, 2003.

[8] P. M. Joshi, and S. Liu, " Web Document Text and Images Extraction using DOM Analysis and Natural Language Processing", ACM, DocEng, 2009.

[9] Y. Li and J. Yang, "A Novel Method to Extract Informative Blocks from Web Pages", IEEE. DOI 10. 1109/ JCAI, 2009.

[10] R. R. Mehta, P. Mitra, and H. Kamick, "Extracting Semantic Structure of Web Documents Using Content and Visual Information", ACM, Chiba, Japan, May 2005.

[11] S. Gupta, G. Kaiser, D. Neistadt, and P. Grimm,"DOM-based Content Extraction of HTML Documents", Pro. 12 th International Conference on WWW, ISBN: 1-58113-680-3, 2003.

HUDSON PLUGIN DEVELOPMENT

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Abstract- Hudson is the excellent open source CI (continuous integration) server. Integral to Hudson is a plugin mechanism that allows third-parties to extend Hudson in several ways, including reporting test failures, results of automated code inspections, notification of broken builds and publication of build artefacts. Hudsonplug-in development is concerned with SCM (Software Configuration Management). Software configuration management is a software-engineering discipline comprising the tools and techniques (processes or methodology) that a company uses to manage change to its software assets.SCM is the process that defines how to control and manage change. The need for an SCM process is acutely felt when there are many developers and many versions of the software. In a complex scenario where bug fixing should happen on multiple production systems and enhancements must be continued on the main code base, SCM acts as the backbone which can make this happen.Hence the goal is to develop plugin that will add extra functionalities to the Hudson Software and enable customizing the functionality of an application.

***Keywords-** Hudson, SCM,CI, Plug-in*

I. INTRODUCTION

A. ContinuousIntegration

With every commit performed by the individual developer to the source code brings about changes that have to be tested and integrated with the base software product. These commits are performed by one of the many individuals working on the project at the same time or separately. Each developer works on the base project to which changes have to be added. Whilst working, he needs to have a stable, latest and integrated version of the product.

Continuous integration is a practice which helps to achieve this by performing the unit testing followed by the integration testing. Continuous integration involves frequently integrating one's new or changed code with the existing code repository. Also the commit should occur so frequently that there should be no intervening window between the code commit and the product build. This helps the final product to be integrated without the chaos leading to integration hell.With continuous integration, no one needs to drop everything and run a release build, these builds are generated everysingle day, and in the most mature environments, a fully tested and verified system can be deployed to productionat any time. In other words, when you automate build, test, and verify using a tool like Hudson you can continue developing our applications without having to wait (or synchronize) on some manual build, test and verify process.

B. Plugin Integration:

Steps for Hudson Plugin Development

Step 1: Generate the plugin skeleton

1.1.1 In a terminal, type the command mvnhpi: create This command tells maven to create the necessary sources. Wait until maven downloads all the required jars and plugin to execute the command.

1.1.2 When prompted provide:

Enter the groupId of your plugin: org.sample.hudson

Enter the artifactId of your plugin: javaone-sample

When maven successfully creates the Hudson plugin, the message "BUILD SUCCESS" will be printed.

Step 2: Build the plugin project

1.2.1 Build the project using maven cdjavaone-sample mvn package The package command tells maven to build the project and create the plugin packages needed by the Hudson server.On successful build of the plugin project,the message "BUILD SUCCESS" will be printed.

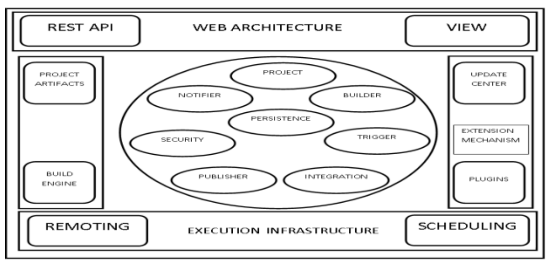
Step 3: Open the plugin project using Eclipse

Use Eclipse IDE to open the project.

Step 4: Run the Plugin Project

In this step we run the project and see the result of the extension added by plugin.

II. ARCHITECTURE



The service objects of Hudson are Model Objects that are run able. Hudson executor runs these services to complete an execution.Model objects are building blocks of Hudson Platform. They hold the data and state of a Job/Run. Hudson provides a security mechanism which allows Hudson Administrators to control areas of access to users or group of users. The key definitions are:

1) Authentication- Determines the identity of a user or roles and represents the Authenticated mode of the user or roles.
2) Authorization - Determines what resources can be accessed or what actions can be executed by an authenticated user or process.
3) Role- Represents a set of functional responsibilities and specific permissions. Users and groups are assigned to roles to authorize thesepermissions.

Each Hudson page is rendered from a Model object and associated Page Definition. The page definitions are stored in Jelly files and written using jelly tags.

The page is constructed from
1) Header - Displays Breadcrumb and user related action links.
2) Side Bar with two sections
3) Action links corresponding to that Dashboard
4) Ajax based sub view to display Build History or Executor status
5). Content Pane where the dashboard specific contents are displayed
6) Footer where the copyright related material are displayed.

The entire execution system can be broadly divided in to the following components.

1) Execution Service: Does the actual computation in a master-slave configuration.
2) Queuing: Responsible for queuing the jobs anddoes the scheduling logic.
3) Job Type: Defines the job types and run types.
4) Load Balancer: Responsible for channeling the queued jobs to the right execution components.
5) Scheduling logic:After the Job is submitted, it goes through the following states, before picked up by a Executor for Execution.
6) Waiting List:Items that cannot be built yet, waiting for queue maintenance.
7) Blocked Job:Can be built, but blocked due to various reasons depending onupstream project or resource.
8) Buildable Job:Promoted to buildable and will be start building as soon as anExecutor becomes available.
9) Pending: Got the Executor, but in pending state due to quiet period etc.
10) Run state:When the job is at run state, it can be scheduled again.

Hudson Extension can be achieved through two modes: Update center and Plugin installation. Hudson provides timely updates which includes security updates, UI updates, configuration updates. The other extension mechanism is through the plug-ins. Hudson has a very active plugin development community which provides a large number of plugins. Also one can easily develop his own plugin according to the requirement.

III. PROPOSED WORK

A. Display label of each build slave on main dashboard: Component gets build on each dedicated build slave. Each slave is part of tag (label), which gets referred when build gets fired. As of now, there is no provision to display tag of build slave on main dashboard provided by Hudson. To get this detail, we need to go to configuration of job. Hudson dashboard needs to be extended to display tag information of each slave

B. Permission based configuration option: Hudson Job Configuration option allows user to modify the settings of each buildable component. This option normally holds with Administrator. Requirement here is to extend this option to other non-Administrator users based on permission checks. This will allow such users to modify the configuration to suffice their requirement.

C. Pause the build whenever required: Need mechanism to pause the build at a certain stage.

D. Combo-box option in parameterized builds: Hudson configuration page currently supports check-boxes for various parameters. This leads to selecting inappropriate combination of parameters. Need to allow combo-box based selection of parameters.

E. Select build slave at runtime using string and regular expression: Currently, Hudson jobs are tightly coupled with build-slaves. This needs to be changed so as to have loose coupling between jobs and build slaves such that if assigned build slave is unavailable, the job should be fired on secondary build slave. The slaves should be selected using a string expression or a regular expression.

F. Multiple perforce connection: Hudson enables single perforce connection. For complex systems, more than one connection is needed.

G. Display trend data whenever required: Hudson reporting is limited and does not provide comprehensive analysis on trend data. Need to extend Hudson reporting to provide comprehensive trend data which will help analyze the state of build processes.

IV. IMPLEMENTATION

A. Purpose:

Component gets build on each dedicated build slave. Each slave is part of tag (label), which gets referred when build gets fired. As of now, there is no provision to display tag of build slave on main dashboard provided by Hudson. To get this detail, we need to go to configuration of job. Hudson dashboard needs to be extended to display tag information of each slave.

B. Algorithm:

1) start
2) add package org.hudsonci.plugins
3) importnode_monitors package
4) create class DisplayNodeLabels and extend NodeMonitor package
5) include class AbstractNodeMonitorDescriptor from node_monitors
6) create object Descriptor to display the labels on the main dashboad
7) if slaves are active
8) usegetNode() function to retrieve node name
9) usegetLabelString() to retrieve corresponding label
10) end

node_monitors is one of the packages that contains classes related to columns on dashboard.
Descriptor is used to extend the code throughout the Hudson.
AbstractNodeMonitorDescriptor gives name for the column.

V. COMPETITIVE ANALYSIS WITH ANOTHER TOOL

ISSUE	CRUISE CONTROL	HUDSON
Installation	Straightforward: unpack a zip of files and run the startup script.	Trivial: a single executable jarfile is provided. Can also run in a servlet container.
Upgrade	Releases are infrequent but require manual upgrade.	Minor releases are frequent but easy to accommodate just by replacing a single file and then restarting the server. Automated upgrade can be launched from the web UI.
Container technology	Comes with Jetty.	Comes with Winstone. Can also run on another standard servlet container such as Jetty or Tomcat5.
Stability	Good - but prone to certain types of build failure, e.g. deadlocks have caused problems.	Very good
Dashboard security	None; all users can alter the small range of editable parameters but most configuration is done through XML.	User authentication is via HTTP Basic mechanism. Credentials are checked against a local table or a configurable LDAP service via the servlet container.
Build history management	There are no tools to manage old builds. Deletion or archival of old builds has to be done manually using the Linux command line.	Individual builds can be either deleted if unwanted, or marked as permanent if needing keeping.

VI. ADVANTAGES

A. Extensibility:
The application can be dynamically extended to include new features.

B. Parallel development:
Since features can be implemented as separate components, they can be developed in parallel by different teams.

C. Clear development direction:
Since the plugin framework ideally provides a well-defined interface and documentation for plugin writers, developers have a clear roadmap for development.

D. Simplicity:
A plugin typically has one function, and so developers have a single focus

VII. CONCLUSION

The current version of Hudson software does not provide the above mentioned 7 features. So at the end of this project an extended version of Hudson software featuring the above characteristics will be hosted at the official Hudson website (http://hudson-ci.org/)

REFERENCES

[1] http://hudson-ci.org/

[2] http://wiki.eclipse.org/Hudson-ci/writing-first-hudson-lugin

[3] http://wiki.eclipse.org/Hudson-ci/Meet_Hudson

[4] http://wiki.eclipse.org/Hudson-ci/Use_Hudson

[5] http://wiki.hudson-ci.org/display/HUDSON/Plug-in+architecture

[6] https://github.com/hudson

[7] https://github.com/hudson-plugins

HEALTHCARE SERVICES USING MOBILE COMPUTING IN ANDROID

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Abstract- Healthcare system using mobile computing with the help of medical expert system provides advantages to patients, enabling them to access medical information and support systems, independent of their current place and time. This paper describes a tool which improves the quality of treatment for patients using mobile application. Our application, iCare, runs on several Android based devices with 3G and Wi-Fi network capabilities. This application is cost effective for patients since the patient don't need to be present physically. This application framework provides support for terminal and application mobility and enables easier implementation. This application accepts the symptoms from the patients, processes the data, identifies the particular disease and hence provides appropriate medication using medical expert system by pattern matching techniques. The data collected from the device is evaluated using expert system techniques which estimate the probability of severity of the disease. As patient uses this application more, expert system will better learn and identify various patterns of diseases which increase the accuracy of iCare.

I. INTRODUCTION

Mobile healthcare applications receive more and more attention due to the ability to reshape healthcare delivery, for example, enabling self management of patients while they pursue their daily activity .Mobile healthcare web services using Android can provide advantages to patients, enabling them to query their Symptoms and get the expert response from the Expert System in the form of identification of the disease and medications to cure the illness. Patients can access medical information and Expert system independent of their current place and time and content can be dynamically adjusted to the current context and terminal type. Mobile devices, home computers and embedded patient terminals can be utilized in healthcare services to provide delivery of information to patients at the point of need. In this way, patients can be equipped with powerful tools and support Systems that can help them in their everyday health management and patients can get more involved in decision making regarding their own health. Problems related to increasing healthcare costs and the higher demand for healthcare personnel and services can be addressed and reduced.

The presented system, iCare, is designed to partially alleviate stress, financial burden, and workload on the doctor.

It is an application which runs on any Android version. Additionally, for healthcare applications and services usability, user-friendliness, and usefulness of the system are very important due to great spectrum of potential future users and variety of their needs and expectations from system's functionalities. Adaptation of interface elements and user interaction to different contexts of use (e.g., types of devices and their characteristics, OSs, and communication - network types) is primary requirement. The way in which mobility issues (e.g., session transfer, handoff between networks) are managed can also greatly influence usability of the system due to additional requirements for user interaction and adaptation of interface for a new context of use Application mobility as defined in enables a user to start interacting with the service using one device and transfers the session to another device automatically when a new communication channel is opened. While the session is transferred, an additional mechanism to warrant session consistency must be provided to protect storage and to avoid transfer of invalid session data. Application mobility must also be provided within an acceptable time frame. Due to the different network types that can be utilized, it is highly possible that communication can be interrupted unexpectedly and that a system must be able to save current session data and enable a user to continue a previously started session without information loss.

When the application is resumed on the same terminal over a different network or on a new terminal, a new authentication process must protect security. Because this may happen frequently, it is necessary that the renewed authentication is both user-friendly and secure. Expert systems are being increasingly used in medical environments such as hospitals, laboratories, and intensive care units, with a view to improving the quality of health care and reducing the likelihood of incorrect medical decisions. Transformation of these systems into mobile solutions would extend their benefits and facilitate their integration into medical environments.

New generations of mobile devices offer users new modes of interaction with medical expert systems.

Our approach involves the use of a mobile client-server model employing web services in order to transfer the currently available web-based system onto an Android platform. The server is dedicated to provide an interpretive report of the obtained test results, whereas the client acts as a convenient user front-end. Communication between the client and the server is based on web services.

The patients have their user accounts on the patient support system (iCare) and can access the service from remote places or their homes, cars or offices using different types of networks (e.g. Wi-Fi, GPRS, 3G).

II. METHODS AND DESIGN

A. Software

This application is designed on the Android software stack produced by Google. Android is an open source framework designed for mobile devices. It packages an operating system, middleware, and key programs. The Android SDK provides libraries needed to interface with the hardware at a high level and make/deploy Android applications. Application is written in Java and use SQL databases to store persistent data. We choose this platform as opposed to others because of the ability to easily thread background running processes, the polished Navigation API, and compatibility with other Android devices. Unlike dedicated systems this software is intended to integrate with the device's existing applications; iCare must share resources with other applications. To make for a pleasant integration, it runs as inconspicuously as possible while using limited resources. Only when the probability of wandering is high will the activity wake up and interrupt the patient. Based on the probability evaluation and patient's response the app can take different actions. Which allows iCare to run harmoniously on the system while minimizing memory consumption and providing ease of use to the patient.

B. Medical Expert System (MES) Design

An expert system is an artificial intelligence application that uses a Knowledge base of human expertise to aid in solving problems. The degree of problem solving is based on the quality of the data and rules obtained from the human expert. Expert systems are designed to perform at a human expert level. In practice, they will perform both well below and well above that of an individual expert. The expert system derives its answers by running the knowledge base through an inference engine, a software program that interacts with the user and processes the results from the rules and data in the knowledge base. Expert systems are used in applications such as medical diagnosis, equipment repair, investment analysis, financial, estate and insurance planning, route scheduling for delivery vehicles, contract bidding, counselling for self-service customers, production control education and training. Tasks such as: monitoring, design, control, simulation, learning support and information retrieval, among others can be done through the use of expert systems. As elaborated in the problem statement/formulation, patients with sicknesses such as minor headaches, minor stomach aches, and minor malaria need not be report to a major referral Hospital such as Ruby Hall Clinic in order to release some amount of work pressure on Medical Doctors. To solve this problem, this paper through mobile technology and cloud computing network technology proposes a Mobile Medical Expert System (MES) that has the knowledge base of diagnosis, advice and treatment of these minor sicknesses mentioned earlier. Patients would initially register as user through an interface and interact with the MES and the Medical Doctor through cloud computing, mobile technology and devices. The Medical Doctor will advice the patient through their mobile devices according to his/her interaction with the MES.

C. Symptoms Identification Mechanism

The proposed System Scenario involves the following steps:

1. Patients logs into the System with mobile device through cloud server, after registration through the Hospital Administrator.

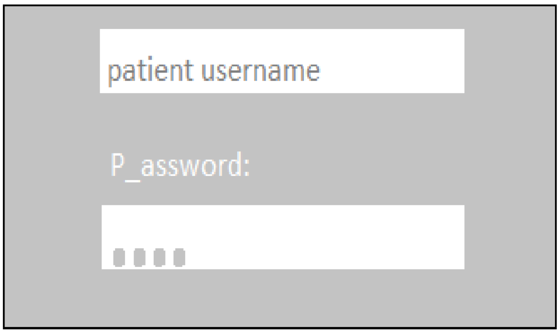


Figure1: iCare Log in Interface

2. After logging in, the patient initially interacts with the Medical Expert System through by a Medical Diagnostics Interface on his/her mobile device.

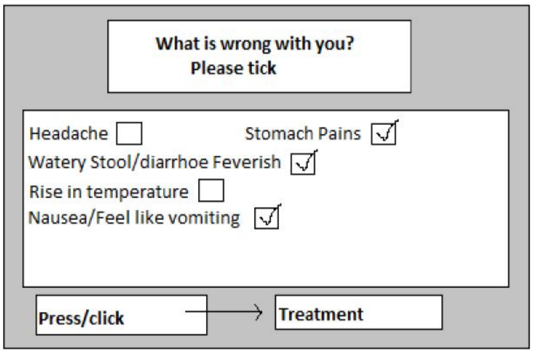


Figure 2: iCare Medical Diagnostic

D. Prescription Mechanism:

1. After clicking of "Treatment", the next interface if the patient's iCare Prescription and Treatment advice.

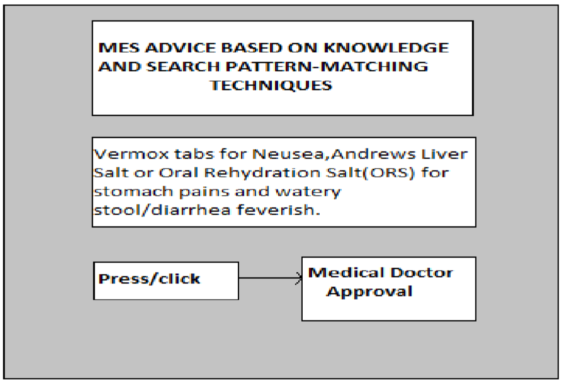


Figure3: iCare Medical Treatment Advice

2. After click Medical Doctor Approval, a query of patient's diagnosis and MES (Medical Expert System) advice is sent to the Medical Doctor's mobile device for approval of MES advice or non approval of MES advice for onward consultation of Medical Doctor physically.

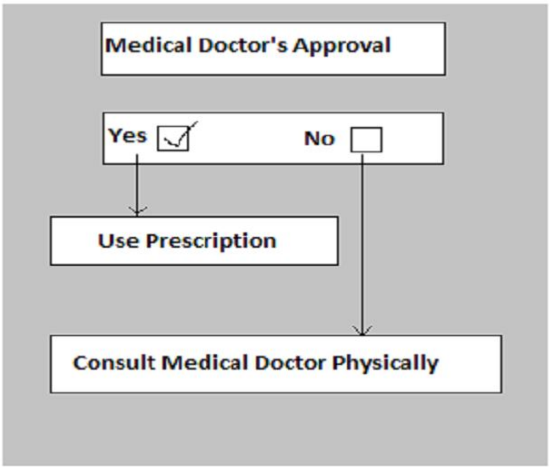


Figure 4: Medical Doctor Approval

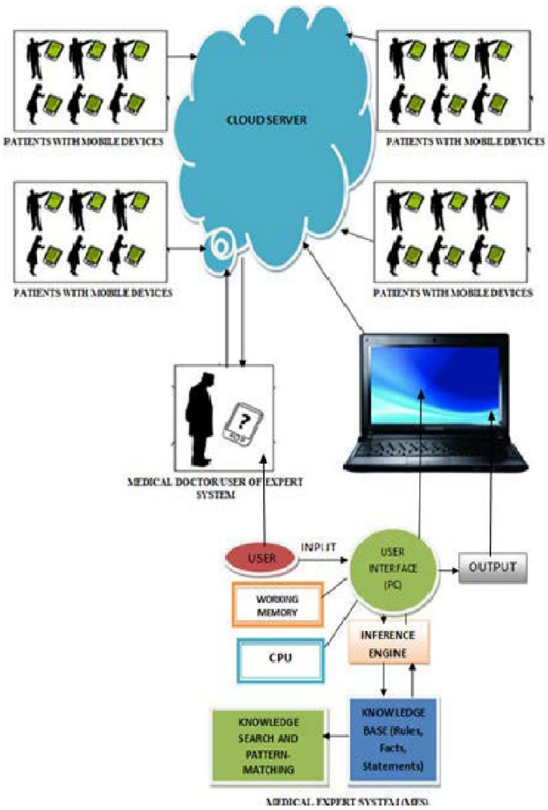
E. Instant Messaging

Instant messages (IM) are short text messages exchanged between users that want to chat in real time. After a user signs on to IM from a mobile device, a list of Doctors (referred to as a Expert list) appears on the mobile subscriber's screen using familiar screen names. The mobile subscriber can send a message to the IM service requesting to see, with the help of special icons, who is online and available to chat. IM messages can be sent only to Doctors that are online.

F. Real-Time Database Download

This system database will be stored on cloud due to this memory required on mobile device will be less. In case of network failure, user can get the medications with the help of Database, that client has previously downloaded in the form of excel sheet. Hence, client can easily access the database in the remote places where network is not available.

III. FLOW DIAGRAM



IV. CONCLUSION

Thus our Application works for the benefits of the society and acts as VIRTUAL DOCTOR for the patients. Our application helps the user to get a modified and advanced version of the famous android application, Common Symptoms Guide. This application provides an interactive interface between the client (patient) and the server (Medical Expert System).

This paper proposed a Medical Expert System (MES) that can be used to solve problems of too many patients seeking daily medical attention. This research showed that some of these patients need not attend a major referral Hospital, because their sicknesses are minor and may not require hospital attendance. The proposed system when implemented will not only reduce patient numbers but also help Medical Doctors to speed up diagnosis and treatment of patients through the advice and interaction with a MES.

REFERENCES

[1] Nana Yaw Asabere, "mMES: A Mobile Medical Expert System for Health Institutions in Ghana," IJST 2012.

[2] Jelena Mirkovic (2012), "A framework for the development of ubiquitous patient support systems".

[3] Schreier G, Hayn D, Ammenwerth E,Expert System Based Interpretation Of Hepatitis Serology Test Results As App Store Iphone Application,".

[4] Francois Andry, Lin Wan and Daren Nicholson(2011)," A Mobile Application Accessing patient's Health Records Through A Rest API how REST-Style Architecture can Help Speed up the Development of "Mobile Health Care Applications".

[5] Frank Sposaro, Justin Danielson, Justin Danielson", iWander : An Android Application for Dementia Patients," IEEE 2010.

[6] Francois Andry, Lin Wan and Daren Nicholson, "A Mobile Application Accessing Patients' Healthcare Records Through a Rest API,"IEEE 2012.

CONVERTING THE SINGLE TENANT APPLICATION TO MULTI TENANT APPLICATION

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Abstract- the concept of multi tenancy is, single instance of software runs on a server and serves multiple client organizations. A software application is designed with a multitenant architecture to virtually divide the data and configuration where each client organization can customize its own virtual application instance. This project aims to demonstrate conversion of single tenant application to multi tenant application. The project is applicable for every open source (java) single tenant application that runs over cloud. As a part of this project, an eclipse plug-in is developed that will assist user to migrate the code and make necessary changes.

Keywords- Tenant, Plug-in, API, Cloud

I. INTRODUCTION

In order to reduce the computing expenses, companies rent the space and processing power on time sharing basis. Cloud Computing is the use of hardware & software resources which are provided over a typical network such as Internet, for the services demanded by Client (tenant). The Cloud can be differentiated as Public (shared) Cloud, Private (enterprise) Cloud and Hybrid Cloud. Public cloud mainly focuses on providing services over the internet with voluntary control over the infrastructure. Public clouds provide services on application level (SaaS), Infrastructure level (IaaS), Platform level (PaaS). A Private Clouds are hosted within an organization as it is the private product of the company whereas hybrid cloud combines and integrates the power of both public and private cloud. Interaction with Cloud Software is enabled through API (Application Programming Interface) which is similar to the user interface that facilitates interaction between humans and computers. Multi-tenancy is a feature in which permits sharing of resources and costs across a large pool of tenants thus allowing for Centralization of Infrastructure, Increasing the peak load capacity and Development in Utilization & efficiency of systems. Fig 1.0 elaborates the idea of multi-tenancy.

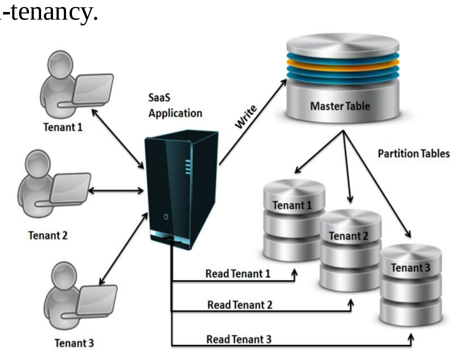

Fig 1.0 Multi-tenant Environment with SaaS Application

II. RELATED WORK

Currently there are some applications that provide multi tenant environment. The Cloud Ninja is one example. The Cloud Ninja Project is a Windows Azure multi-tenant sample application demonstrating metering and automated scaling concepts, as well as some common multi-tenant features such as automated provisioning and federated identity. This sample was developed by the Azure Incubation Team in the Developer & Platform Evangelism group at Microsoft in collaboration with Full Scale 180. The topic of this paper deals with conversion of single tenant application to multi tenant application, therefore, JMeter has been taken as an example to highlight the various scenarios in a Single Tenant Application and to Demonstrate the step by Step procedure for its conversion.

The Apache JMeter desktop application is open source software, a 100% pure Java application designed to load test functional behavior and measure performance. It was originally designed for testing Web Applications but has since expanded to other test functions.Apache JMeter may be used to test performance both on static and dynamic resources (files, Servlets, Perl scripts, Java Objects, Data Bases and Queries, FTP Servers and more). It can be used to simulate a heavy load on a server, network or object to test its strength or to analyze overall performance under different load types. You can use it to make a graphical analysis of performance or to test your server/script/object behavior under heavy concurrent load.

III. PROPOSED APPROACH

For the conversion of a single tenant application to a multi tenant application we are considering a pure java, single tenant, open source application which is of a significant use when deployed over cloud as a multi tenant application. For the same purpose we are developing an eclipse plug-in that will enable the user to migrate the source code. The plug-in will scan the Java code and will generate a report that will specify the code that needs to be modified. It provides separate workspace facilitating to work in different project for users. The plug-ins is developed on Eclipse -Juno platform. In addition a Source code analyzer PMD plug-in is used which finds unused variables, Empty catch blocks, and unnecessary object creation making the code efficient. It assists the source code using PMD providing details to optimize the code in efficient format.

Source code of single tenant application will be loaded in the project explorer of the eclipse Juno. A context menu is added to the project explorer named “Scan for Multi-tenancy”. This menu helps us to point out the portion in the source code that is needed to modify for multi-tenancy. By changing the highlighted portion we can easily convert the single tenant application to multi tenant application. When we click on the developed context menu, the source code is scanned line by line. The system determines the code to be changed for the respective conversion. In such a way, we can determine the part of source code that is needed to be changed.

Along with the context menu, two views are created “multi-tenancy” and “report”. As we check for the multi-tenancy by clicking on the developed context menu, the multi-tenancy view highlights the code that is needed to be changed or modified. The report view generates the lines of code that is to be modified.

We are using Windows Azure as A cloud platform. Windows Azure is Microsoft's application platform for the public cloud. This platform can be used in many different ways. For instance, to build a web application that runs and stores its data in Microsoft data centres or just to store data, with the applications that uses this data running on-premises (that is, outside the public cloud). It can also be used to create virtual machines for development and test or to run SharePoint and other applications. It is primarily used to build massively scalable applications with lots and lots of users. Because the platform offers a wide range of services, all of these things and more are possible.

IV. EXPECTED RESULTS

This eclipse plug-in will high light the code which user will need to change for converting single tenant application to Multi-tenant application. End user will be able to install this plug-in using update site functionality in eclipse. This plug-in will provide a context menu of all dynamic web project saying ‘find changes for Multi-tenancy’. After clicking on this menu, plug-in will check and display result in a separate view.

V. ADVANTAGES OVER SINGLE TENANCY

1. In multi-tenancy, software development and maintenance costs are shared between the tenants.
2. The change made on a server side reflects at all tenants connected to it. This results in faster updating.
3. Each tenant’s data is isolated and remains invisible to other tenants.

VI. CONCLUSION

Most of the single tenant applications face the problem of high cost, efficiency and functionality. In this paper we present and experimentally validate a novel technique for converting a single tenant pure java application based on cloud platform into a multi tenant application by creating a plug-in. This plug-in will direct the client in order to get the application converted into multitenant one by pointing to the java code that needs to be changed for the respective conversion. Such a conversion into multi tenant application will enable a client to use the application in a flexible and secured way along with reduced cost factor and enabling the client to customize its own application.

ACKNOWLEDGEMENT

Authors would like to thank Mr.D.A.Kulkarni for their guidance, Mrs.Gitanjali Thete for her help and support and Persistent Systems Limited for sponsoring this project.

REFERENCES

[1] Venkatesh, M.Sarveswaran, V. Varalakshmi, P. “Client centric multi-tenancy for cloud services”, Chennai, TamilNadu

[2] L. Mac Vittie, “Architectural Multitenancy,” blog,18May2010;https://devcentral.f5.com/blogs/us/architectural-multi-tenancy#.UREwhsWwblY

[3] J. Meiers, “Best Practices for Cloud Computing Multitenancy: Building a Scalable, Available CloudComputing Multi-tenancy Architecture,” IBM,5July2011;www.ibm.com/developerworks/cloud/library/cl-multitenantcloud

[4] Cloud Computing: Web-Based Applications That Change the Way You Work and Collaborate Online, Michael Millar, Edition: 1

AUTHENTICATION FOR ATTACKS IN GRAPHICAL PASSWORDS PASS POINTS STYLE

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Abstract- Access to computer systems is often based on the use of alphanumeric passwords. However users have difficulty in remembering passwords that is long and random-appearing. Graphical passwords have been designed to try to make passwords more memorable and easier for people to use. Using graphical passwords users click on images rather than type alphanumeric characters. We have designed a new and more secure graphical password system called passpoints.Based on click-order patterns we introduced a proposed graph-based algorithms to create dictionaries with focus-of-attention scan paths

Key Terms- *Graphical passwords, passpoints, authentication.*

I. INTRODUCTION

Graphical passwords have been designed to try to make passwords more memorable and easier to use. Graphical passwords are an alternative to text passwords where user is asked to remember an image or parts of an image instead of a word.We are further discussing new and more secure graphical password system called passpoints. In passpoints system users create a fivepoint click sequence on a background image.Complex images an have hundres of memorable points,for ex:with 5 or 6 click points one can make more passwords than 8-character unix-style passwords.Some tolerance region is set for each password because the tolerance [2] gives a margin of error around the click point in which the user's click is recognized as correct. With the help of passpoint-style graphical passwords human seeded attacks have been used by human computed data which is further used to make easier and efficient attacks.The attacker require such attacks to collect the adequate "human computed "data for an image targeted [1],[9].Evaluating and introducing a set of purely automated attacks against the passpint-style graphical passwords.Users are more interested in choosing a click points based on some hypothesis that hoosing a click points in that areas of image where the user attention is accordingly drawn towards i.e,five click points in a straight line is called click order patterns. Dictionaries where generated from the graphical passwords for the use in a dictionary attack. A successful attack must be able to efficiently generate a dictionary containing highly probable passwords. the cost of the dictionary depends on the different background images.

II. LITERATURE REVIEW

2.1 RELATED WORK

We concentrate on click-based graphical password schemes where a user cliks on a number of set points in a background image and work is related to guessing attacks on graphical passwords. In Blonder's proposal users click on a set of predefined regions [3].

In Dhamija and Perrig's proposal user is asked to select a number of images from a set of random pictures.

Passpoints allow users to click a sequene of some points anywhere on an image with a error tolerance.e,error tolerance can be st to as p=4.An attacker ould predict hot spots by using image processing toole for guessing passpoints passwords and for other images their mthod guessed 9.1% and 0.9% pf passwords on two images using an dictionary attack 2^{35} entries compared to password space 2^{43} password.

Some of the click-order patterns evaluated with human seeded attacks is DIAG and other is LINE and other click based graphical password schemes CCP [7] and PCCP [6].

The major advantage of passpoints is its large password space over alphanumeric passwords.The large password space is significant because it reduces the gueesability of passwords

2.2 Survey

Recognition based systems

Recognition based systems also known as cognometric systems or search metric systems generally require that users memorize a portfolio of images during password creation, and then to log in, must recognize their images. Recognition based systems have been proposed using various types of images. Phishing attacks are somewhat more difficult with recognition-based systems because the system must present the correct set of images to the user before password entry. Shoulder-surfing seems to be of particular concern in recognition-based systems when an attacker can record or observe the images selected by users during login [6].

For PassFaces, the analysis of user choice by Davis et al. [10] showed that users tend to select attractive faces of their own race; and that users selected predictable sets of faces such that an attacker knowing one face could determine the face most likely to be selected as the next password part. Because users tend to select predictable images, successful dictionary attacks may be expected, as well as personalized attacks, e.g., if attackers know a user's race or gender. Davis et al. [10] guessed 10% of passwords created by male participants in 2 guesses. A major conclusion was that many graphical password schemes, including Faces, may require "a different posture towards password selection" than text passwords, where selection by the user is the norm. As noted in Section V (which also mentions user choice issues in the Story scheme [10]), a phishing attack on PassFaces requires a MITM attack

Recall based systems

In this section two types of picture password techniques used reproduce a secret drawing and repeat a selection. In these systems, users typically draw their password either on a blank canvas or on a grid i.e, DAS technique proposed by Jermyn [4].

Passlogix [11] has also developed several graphical password techniques based on repeating a sequence of actions. For example, its v-Go includes a graphical password scheme where users can mix up a virtual cocktail and use the combination of ingredients as a password. Other password options include picking a hand at cards or putting together a "meal" in the virtual kitchen. However, this technique only provides a limited password space and there is no easy way to prevent people from picking poor passwords (for example, a full house in cards).

2.2 Graphical password Attacks

Brute Force Attack

This type of attack uses an algorithm that produces every possible combination of words to break the password. Text-based password contains 94^N number of space where 94 is the number of printable characters (including space) and N is the length. This type of attack has always proven successful against text-based password because of its ability to check all possibility within the length of the password. As such, users are advised to select a stronger and complex password to prevent discovery from brute force attack (Eiji Hayashi, 2008). However, GUA proves to be more resistant to brute force attacks since the attack software needs to produce all possible mouse motions to imitate passwords especially when trying to recall the graphical passwords. One of the reasons that helped is the large password space present in most graphical passwords techniques which is not available in the textual variant[5].

Dictionary Attack

This ingenious attack uses words found in the dictionary to check if any were used as passwords by the users. Many users' uses weak passwords which make it easier for attackers to guess the password using the graphical dictionary attack[6]. Because of graphical password method of using mouse input type recognition, using dictionary attack on GUA would be a waste of time. Dictionary attacks against recognition and cued-recall graphical password systems require more effort up-front than against text passwords or recall-based graphical passwords,since attackers must first collect one or more of a set of images.Images gathered for one system will not help attacks on a second system, unless both systems use the same image set.During recall, it is more difficult and complex to use the automated dictionary method to produce all possibility of a single user click of an image than a text-based attack [6-8].

Spyware Attack

This attack uses a small application installed on a user's computer to record sensitive data during mouse movement or key press. This form of malware secretly store these information and then reports back to the attackers system. With a few exceptions, these key-loggers and listening spywares are unproven in identifying mouse movement to crack graphical passwords. Even if the movement is recorded, it is still not accurate in identifying the graphical password. Other information is needed for this type of attack namely window size and position as well as the timing[9].

Shoulder-Surfing Attack

As the name implies, passwords can be identified by looking over a person's shoulder. This kind of attack is more common in crowded areas where it is not uncommon for people to stand behind another queuing at ATM machines. There are also cases where ceiling and wall cameras placed near ATM machines are used to record keyed pin numbers. The best way to avoid pin numbers being recorded or remembered by attackers is to properly shield the keypad when entering the pin number[10-12].

Guessing attack

Since many users try to select their password based on their personal information like the name of their pets, passport number, family name and so on, the attacker also tries to guess passwords by trying these possible passwords. Password guessing attacks can be broadly categorized into online password guessing attacks and offline dictionary attacks. In an online password guessing attack, an attacker tries a guessed password by manipulating the inputs of one or more oracles. In an offline dictionary attack, an attacker exhaustively searches for the password by manipulating the inputs of one or more oracles (Roman, 2007).

III. TERMINOLOGIES

Some of the hypothesis where used to choose a passwords which consists of click points i.e distinguishable points and calculable points with corner detection and centroid detection [1].Corner detection contains intersection of the two edges and centroid detection contains objects in the center[1].

3.1 Dictionary Generation Algorithm

Inputs:permutations,digraph
Output:valid passwords generated

Step 1:subdictionary is generated and is set to Null for(Digraph is set with vertices 1 to n)
Step 2:paths are allocated with some path finding algorithm withi nputs subdictionary, path of length Path is a set of paths of lengths i.e.,passwords
Step 3:subdictionary is a set of combination of paths
Step 4:end of for statement
Step 5:return back to subdictionary

In the above algorithm if a attacker plans to generate a dictionary which may consist of subsets of all the permutations involved which must satisfy the some predefined conditions one proposed approach is to generate only those permutations that satisfy the conditions heuristically.
This algorithm generates a subdictionary for a subheuristic such as right-to-left and left-to-right clikoder pattens.
The advantage of this algorithm is it increased the validity of passwords and long-term memorabiliy.[7]

3.2 Path Finding Algorithm

Inputs: length,source,digraph dg

Outputs:combinations of all subdictionaries is a final attack dictionary

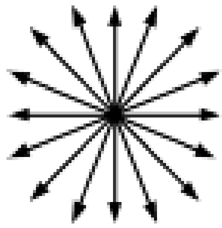
Step 1: subdictionary sd is set to Null Upon termination sd is a set of paths of length
Step 2: for all the nodes nearby neighbor S should belong to the source.
Step 3: neighbors S would be the set of neighbos of node S
Step 4: if the length is less than 1 then paths go back to the step1
Step 5: path prepends source Prepend (paths ,S) pepends node S to path p Paths is set with source and node S
Step 6: sd is the combination of all paths

Finding paths is a recursive function which finds all paths of defined length from the node in digraph From this algorithm the edges can be defined by the points in an image and distance between the image can be measured and the creation time for attacks depending on the number of dictionary entries. Finding paths in an image can be identified by applying kruskal’s algorithm as a proposed method o some more algorithms.

3.3 Different Click-Order patterns

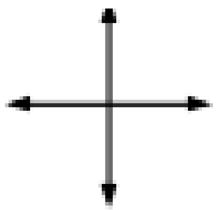
DIAG pattern

DIAG is a lazy variation.It includes sequence of any five click-points that can be in horizontal and vertical direction.DIAG dictionary is a set of four union sets of passwords containing right to left,left to right,top to bottom,bottom to top.



LINE pattern

LINE pattern is a super-lazy variation.It includes sequence of any five click-points that can be in horizontal and vertical line



IV. PASSPOINT METHOD

To improve upon the shortcomings of the Blonder Algorithm, in 2005, PassPoint was created Passpoint wasable to fill in the gaps left by blonder. In this case the image could be any natural picture or painting as well as rich enough so as to have several possible click points. Apart from this the image is not secret and has no other role other than that of assisting the user to remember theclick point. Furthermore it is not as rigid as the blonder algorithm which requires the setting of artificial predefined click regions with well-marked boundaries.[8] The authentication process involves the user selecting several points on picture in a particular order. When logging in, the user is supposed to click close to the selected click points, within some (adjustable) tolerance distance, for instance within 0.25 cm from the actual click point Studies indicate that when using the PassPoint system users were easily able to quickly create a valid password. They found it much harder to know their passwords compared to alphanumeric users, hence they had to take a lot more trials and more time to complete the process. Comparatively the login time, in this method is longer than that of the alphanumeric method

V. CONCLUSION

Finally, purely automated attacks are arguably much easier for an attacker to prepare especially if large image datasets are used. corners and centroids of images might be extracted, and used to build a click-order heuristic graph. Finally, our attacks could be used to help inform more secure design choices in implementing Pass Points-style graphical passwords. Pass-points passwords are most robust than text passwords against multiple password interference.

REFERENCES

[1] Paul C. van Oorschot, Amirali Salehi-Abari, and Julie Thorpe “Purely Automated Attacks on PassPoints-Style Graphical Passwords”

[2] S. Wiedenbeck, J. Waters, J. C. Birget, A. Brodskiy, and N.Memon, “Authentication using graphical passwords: Basic results,”in Proc. Human-Computer Interaction Int. (HCII), Las Vegas, NV, 2005.

[3] G. Blonder, “Graphical Passwords,” U.S. Patent 5559961, 1996.

[4] X. Suo, Y. Zhu, and G. S. Owen, “Graphical passwords: A survey,” in Proc. 21st Annu. Computer Security Applications Conf. (ACSAC), Tucson, AZ, 2005.

[5] Hayashi, E. and N. Christin, Use Your Illusion: Secure Authentication Usable Anywhere, in Proceedings of the 4th symposium on Usable privacy and security (SOUPS). 2008, ACM.

[6] Chiasson, S., et al., Multiple Password Interference in Text Passwords and Click-Based Graphical Passwords. ACM,2009

[7] Sonia Chiasson, P.C. van Oorschot, and Robert Biddle” Graphical Password Authentication Using Cued Click Points”

[8] Arash Habibi Lashkari, Samaneh Farmand1”A new algorithm on Graphical User Authentication(GUA) based on multi-line grids

[9] S. Chiasson, P. C. van Oorschot, and R. Biddle, “A second look at the usability of click-based graphical passwords,” in Proc. 3rd Symp.Usable Privacy and Security (SOUPS), Pittsburgh, PA, 2007.

[10] D. Davis, F. Monrose, and M. Reiter, “On user choice in graphical password schemes,” in 13th USENIX Security Symposium, 2004.

[11] Passlogix, "www.passlogix.com," last accessed in June 2005.

COMPARATIVE STUDY OF ALGORITHMS IN GRID COMPUTING USING THREE ALGORITHMS: ANT COLONY OPTIMIZATION, ARTIFICIAL FISH SWARM ALGORITHM, FINE GRAINED ALGORITHM

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Abstract- This paper discusses Grid scheduling algorithms which help in managing heterogeneous resources and in reduction of scheduling computational resources times. These algorithms support for dynamic, reconfigurable on demand, secure and highly customizable computing storage and networking environments. Three scheduling algorithms namely Artificial Fish Swarm Algorithm, Fine-grained Job Scheduling algorithm and Ant Colony Algorithm are mentioned in detail. The comparative mechanisms and concepts of the three scheduling algorithms mentioned above have been presented.

Index Terms- *Ant Colony Optimization, Artificial Fish Swarm Algorithm, Fine Grained Algorithm*

I. INTRODUCTION

Computational Grids are emerging as a new computing paradigm for solving grand challenge applications in science, engineering and economics. Grid development involves efficient management of heterogeneous, geographically distributed, and dynamically available resources[1].

Conceptually a grid is a collection of connected computing resources that appear to users as a unique large system, providing a single point of access to the datacenter. A grid system coordinates resource sharing in a de-centralized manner and uses standard, open, general purpose protocols and interfaces. It delivers non-trivial qualities of service such as guaranteed bandwidth for application, CPU cycles and latency. Grid computing can be described as the coordinated, transparent, and secure sharing of IT resources across geographically distributed sites based on accepted computing standards**.**

Examples of grid implementations are found in computational science, large-scale simulation, data mining, network-enabled solvers and collaborative engineering projects. Grid computing also reflects changing ideas about the nature of software, who owns it, and where it resides. It implements a service based software model in which computing services are configured to meet a specific set of requirements at a point in time, executed, and discarded, approaching the vision of instant computing. Broad adoption of grid computing depends on solving technical and economic concerns, including end-to-end security and usage metering. These issues will initially limit the reach of grids to the boundaries of an organization. In this environment the scheduler becomes one of the most critical components of grid middleware as it has the responsibility of selecting resources and scheduling jobs in such a way that user or application requirements are met in terms of overall time of execution (performance) and cost of resources utilized.

II. GRID SCHEDULING

Mapping of jobs to resources as per their requirements and properties. It is proposed to solve large complex problems. Grid scheduling is an intelligent algorithm capable of finding the optimal resource for processing a job.

Major function is to find the optimal resources and to allocate them to each individual job. The objectives of a grid scheduler are overcoming heterogeneousness of computing resources, maximizing overall system performance, such as high resource, utilization rate and supporting various computing intensive applications, such as batch jobs and parallel applications. The grid scheduler is mainly concerned with the following: CPU utilization – to keep the CPU as busy as possible, throughput - number of process that complete their execution per time unit , turnaround time - amount of time to execute a particular process and response time - amount of time it takes from when a request was submitted until the first response is produced.

III. JOB SCHEDULING ALGORITHM

The artificial fish swarm algorithm (AFSA) is an algorithm based on population evolution similar to the genetic algorithm (GA). The basic argument of AFSA is that in a water area, the place where there are most fish is where there is most food. The optimal search imitates the behaviors of fish to seek food. Every artificial fish is an entity which the self-data information and several self-behaviors are embedded. Its behavior in the next time depends on its current status and the Environmental status, for example the food density and the crowding index. Meanwhile its self-behavior affects not only the environment, but also behaviors of the other fish. The AFSA is implemented by the following four main behaviors of an artificial fish.

Fine-grained Job Scheduling Algorithm (AFJS)- Sending some fine-grained jobs to a resource that can support high processing capability is not economical compared with sending a coarse-grained job. Because the overall processing time for each job includes job scheduling time, job transmission time to a grid resource, job executing time, and transmission time of output, fine-grained jobs spend too much time in scheduling and transmission. Meanwhile, it is wasteful to process a lightweight job with a highly capable computer. The total overhead of fine-grained jobs scheduling can be reduced by grouping the lightweight jobs during the scheduling process for deployment over the grid resources. This algorithm mainly focuses on fine-grained jobs scheduling in a grid, how they are grouped into coarse-grained jobs, and how they are allocated.

Ant algorithm is a new heuristic algorithm; it is based on the behavior of real ants. When the blind insects, such **as** ants look for food, every moving ant lays some pheromone on the path, then the pheromone **on** shorter path will be increased quickly, the quantity of pheromone on every path will effect the possibility of other ants to select path. At last all the ants will choose the shortest path.

IV. ALGORITHMS

A. Artificial Fish Swarm Algorithm

The basic idea of AFSA is to imitate the fish behaviors such as preying, swarming, following with local search of fish individual for reaching the global optimum; it is random and parallel search algorithm. The AFSA is generated from long observation of fish swarm in nature, using the swarm intelligence in the solution of the optimization problem, and combining with the artificial intelligence.

BEHAVIOUR [2,3]:

Artificial Fish (AF) is a fictitious entity of true fish, which is used to carry on the analysis and explanation of problem, and can be realized by using animal ecology concept. With the aid of the object-oriented analytical method, the artificial fish as an entity encapsulated with one’s own data and a series of behaviors, which can accept amazing information of environment by sense organs, and do stimulant reaction by the control of tail and fin.The environment in which the artificial fish lives is mainly the solution space and the states of other artificial fish. The AF realizes external perception by its vision shown in Fig.1 X is the current state of an AF, Visual is the visual distance, and Xv is the visual position at some moment. If the state at the visual position is better than the current state, it goes forward a step in this direction, and arrives in the Xnext state; otherwise, continues an inspecting tour in the vision. The greater number of inspecting tour the AF does, the more knowledge about overall states of the vision the AF obtains. Certainly, it does not need to travel throughout complex or infinite states, which is helpful to find the global optimum by allowing certain local optimum and some uncertainty.

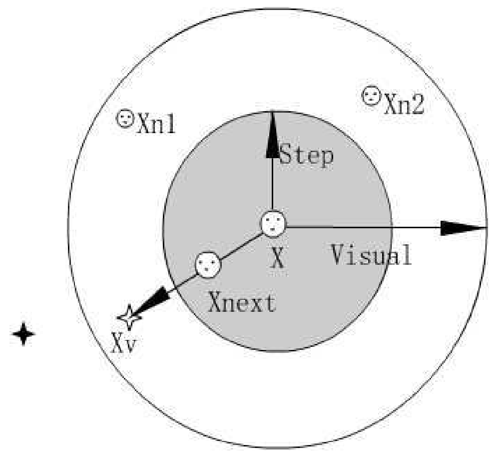

Figure 1 Vision concept of the Artificial Fish.

Let X=(x1, x2,…, xn) and Xv=(x1v, x2v, …xnv), then process can be expressed as follows:

The AF model includes two parts (variables and functions). The variables include: X is the current position of the AF, Step is the moving step length, Visual represents the visual distance, try number is the try number and ∂ is the crowd factor ($0< \partial < 1$). The functions include the behaviors of the AF: AF_Prey, AF_Swarm, AF_Follow, AF_Move.

THE BASIC BEHAVIORS OF AFSA

Fish usually stay in the place with a lot of food, so we simulate the behaviors of fish based on this characteristic to find the global optimum, which is the basic idea of the AFSA. The basic behaviors of AF are defined as follows for maximum [2]:

(1). AF_Prey: This is a basic biological behavior that tends to the food; generally the fish perceives the concentration of food in water to determine the movement by vision or sense and then chooses the tendency.

(2). AF_Swarm*:* The fish will assemble in groups naturally in the moving process, which is a kind of living habits in order to guarantee the existence of the colony and avoid dangers.

(3). AF_Follow: In the moving process of the fish swarm, when a single fish or several ones find food, the neighborhood partners will trail and reach the food quickly.

(4). AF_Move*:* Fish swim randomly in water; in fact, they are seeking food or companions in larger ranges.

THE MODIFIED ARTIFICIAL FISH-SWARM ALGORITHM [3]

(A) THE LEAPING BEHAVIOR

The preying behavior, swarming behavior and following behavior are all local behaviors in some degree. If the objective functions value is not changed after several iterations, it manifests that the function might fall into local minimum. If the program continues iteration, every AF's result will gradually be same and the probability of leaping out local optimum will be smaller. To increase the probability to leap out local optimum and attain global optimum, we attempt to add leaping behavior to AF. The AF's leaping behavior is defined as follow. AF_Leap: If the objective function is almost the same or difference of the objective functions is smaller than a proportion during the given iterations, Chooses some fish randomly in the whole fish swarm, and set parameters randomly to the selected AF.

(B) ADAPTIVE STEP LENGTH IN MAFSA

M.Jian et al, consider the parameter Step, with the increase of the Step, the speed of convergence is accelerated. However, when the increase of the Step is out of a range, the speed of convergence is decelerated, and sometimes the emergence of vibration can influence the speed of convergence greatly. Using the adaptive step may prevent the emergence of vibration, increase the convergence speed and enhance the optimization precision. In the behaviors of AF_Prey, AF_Swarm and AF_Follow, which use the Step parameter in every iteration, the optimized variables (vector) have the various quantity of Step*Rand ( ) , Step is a fixed parameter, Rand() is a uniformly distributed function. Using the adaptive step, we can select the *Step* more randomly which can guarantee the fast convergence, the result's precision and stability.

B. FINE–GRAINED JOB SCHEDULING ALGORITHM

Definition of a Fine-Grained Algorithm [6]:

In grid omputing, we use MI as the unit of jobs. MI is million instructions or processing requirements of a user job. If the MI of a job is less than a fixed threshold max *M*, we consider it as a fine-grained job. And it will be scheduled by fine-grained job scheduling algorithm.

RESOURCE MONITORING [5]

From fig.2 Resource characteristics on which the grouping strategy is based are obtained from mediators. The updated status of resource information is activity transferred from resources to the mediator in push_ based model.

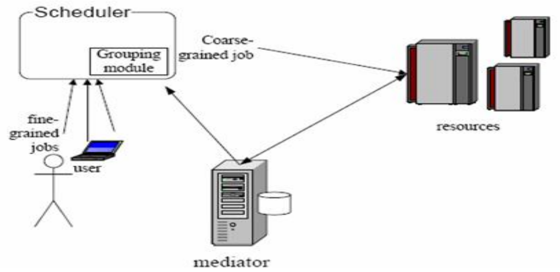

Fig 2: Mechanism for resource monitoring

Algorithm:
Begin
Phase 1: Initialization

1. The scheduler receives jobs from users.
2. The scheduler gets resources status from mediator.
3. According to the MI of jobs, job_list is sorted in descending order using bucket sort algorithm. And each job is assigned a new ID.
4. Define constraint conditions value $Con_{i,j}$ *is*

$(groupedjob\_MI_j + job_i <= MIPS_i * T_{comp\_j}$,and
$(groupedjob\_file\_size + job\_file\_size_i)$
$/baud\_rate < T_{comp\_i})$
$(groupedjob\_MIj=0,$
$Job/MIPS_j > job\_file\_size/baud\_rate)$

Phase 2: Job Scheduling and Deployment

Phase 2 is the process of job scheduling and deployment. In phase 2, given a list of resource, a match grouped job will send to the appropriate resource. First, we group the fine-grained jobs one by one until the first job that didn't satisfy the conditions appear. Then we use binary search to find the next fine-grained job join the grouped job. The job we found is the job that satisfies constraint conditions with the maximal MI

In the phase, when there is no more resource left, first come first served (FCFS) algorithm is used, once a resource node finishes its job, the mediator will get the status of the idle resource, and it will be assigned a new grouped job that grouped By AFJS. Compared with the round-robin algorithm used in , this algorithm reflects the dynamic grid environment more suitably especially when new resources are discovered constantly by resource manager.

Phase 3: Checks if jobs are remaining or not.
This phase checks that all jobs are scheduled or not.If any job is remaining it will check job list if it is not zero then again go for Phase 2 for allocating recourses to particular job. Then, the scheduler

receives computed grouped jobs from resources and split the output before presenting to the user.

C. ANT COLONY ALGORITHM

Ant colony algorithms were inspired by the observation of real ant colonies. Ants are social insects, that is, insects that live in colonies and whose behavior is directed more to the survival of the colony as a whole than to that of a single individual component of the colony. An important and interesting behavior of ant colonies is their foraging behavior, and, in particular, how ants can find shortest paths between food sources and their nest.

While walking from food sources to the nest and vice versa, ants deposit on the ground a substance called pheromone, forming in this way a pheromone trail. When more paths are available from the nest to a food source, a colony of ants may be able to exploit the pheromone trails left by the individual ants to discover the shortest path from the nest to the food source and back. It is also interesting to note that ants can perform this specific behavior using a simple form of indirect communication mediated by pheromone laying , known as stigmergy.

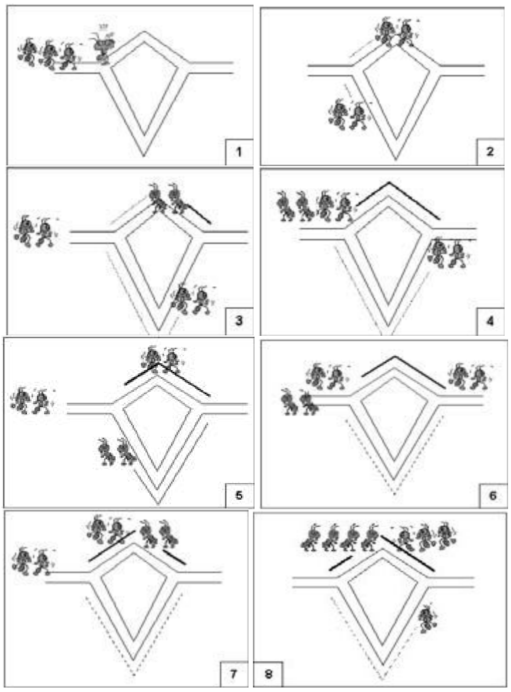

Figure 3: Simulation Evolution Carried out by ant

In order to allow ants to forget poor information, each pheromone value is also decayed at this stage, this is implemented with a parameter ρ which takes a value between 0 and 1[7, 8]. If ρ is set to one then no decay will take place , if ρ is zero then each pheromone value will be wiped at each iteration and the pheromone trail is effectively switched off. Pheromone updating rule is given by:

$$\tau ij(t)new=[\tau ij(t)old]+[\{\rho\}*\Delta\tau ij(t)] \qquad \ldots(1)$$

Where $\tau ij(t)$ →Trail intensity of the edge(i,j).
ρ →Evaporation rate. $\Delta\tau ij(t)$→ Additional pheromone when job moves from scheduler to resource.

The ants usually build a solution using both the information stored in the pheromone trail and the heuristic function. The ant solution building technique is an attempt to follow the concept of the best heuristic method. Each ant starts with an empty schedule and the processor pij best which will complete each unscheduled job j1, j2, j3…. jn earliest is established. A job j is then probabilistically chosen to schedule next based on the pheromone value between j and its best processor and heuristic value. The probability of selecting job j to schedule next is given by the following equation (2). In equation (2), α is a parameter which defines the relative weighting given to the pheromone information, and β defines the relative weighting given to the heuristic information. If α is set to zero then only heuristic information is used and the ants effectively perform a probabilistic search. If β is set to zero only pheromone information is used. The probability selection is given as[8]

$$P_{ij}(t)^k = [\tau\ ij(t)]^{\alpha}*[\eta_{ij}(t)]^{\beta}/\Sigma_{u\in allowed(k)}\ \tau\ iu(t)]^{\alpha}*[\eta_{iu}(t)]^{\beta} \quad \ldots\ldots\ldots\ldots\ (2)$$

Where
$P_{ij}(t)$ →Probability to move along the path (i→j).
$\tau_{ij}(t)]$ →Trail intensity of the edge(i,j).
$\eta_{ij}(t)$ → Visibility (1 / $distance_{ij}$).

The chosen job is then allocated to the best selected ant of each iteration. This process is repeated until all jobs have been scheduled and a complete solution has been built. Each ant in the colony builds a solution in this manner in each iteration. Once all the ants have built a solution the pheromone trail update procedure is performed as described above. It was observed in the test runs that the ants often take some time to start building good solutions because it takes a few iterations before the pheromone trail is populated with good job-processor pairings. After that, ant systems where algorithmically enunciated for optimization in problems like the salesman traveler and others. Ants are social beings with high structured colonies based on very simple individual behavior. Ants smell pheromone and when choosing their way, they tend in probability to the paths marked with stronger pheromone concentrations. When the time pass the pheromone concentration decrease. Repeating same behavior they compose optimized trails that are dynamically defining and they use to find food sources and their nest.

PROPOSED MODIFIED ANT COLONY ALGORITHM [9]

The proposed ant colony optimization is used to solve large complex problems. It requires grid scheduling to achieve high performance. Scheduling of independent jobs remains as a complex problem in grid environment. Hence better scheduling in grid systems can be achieved using heuristic approaches. The Ant colony algorithm – one of the popular heuristic approaches can be used. The basic Ant algorithm involves Transition Probability and Pheromone Updating Rule. Improved ant colony algorithm is, modified ant colony algorithm, used to achieve better scheduling to improve the performance of grid system. The modified ant colony algorithm has changed the basic Pheromone updating rule of original ant colony algorithm. The improved pheromone updating rule is given by:

$\tau_{ij}(t)new=[\{(1-)/(1+\rho)\}*\tau_{ij}(t)old]+[\{\rho/(1+\rho)\}*\Delta\tau_{ij}(t)]$ (3)

Where

$\tau_{ij}(t) \rightarrow$ Trail intensity of the edge(i,j).

$\rho \rightarrow$ Evaporation rate.

$\Delta\tau_{ij}(t) \rightarrow$ Additional pheromone when job moves from scheduler to resource.

V. COMPARITIVE STUDY OF ALGORITHMS

a)

Fine Grained Job Scheduling Algorithm focuses on light weight jobs in grid computing

Artificial Fish Swarm Algorithm solves the optimization problem in signal processing and multiuser detection in communication

Ant Colony Optimization Algorithm solves computational problem further reduced to finding an optimal path in a graph.

b)

Main goal of Fine Grained Job Scheduling Algorithm is to improve the job processing time

Artificial Fish Swarm Algorithm finds global optimum solution.

Ant Colony Optimization search an optimal path in a graph

c)

Fine Grained Job Scheduling Algorithm is based on grouping the jobs into fine grained jobs

Artificial Fish Swarm Algorithm is based on animal ecology concept that is on Artificial fish

Ant Colony Optimization is based on behavior of ants seeking a path between their colony and source of food.

d) Definition:

Light weight job: It has few lines of code or very simple arithmetic expressions

Artificial fish: It is an entity encapsulated with its own data and series of behavior

Ant colony: Their foraging behaviour and by using pheromone, how they find shortest path between food source to their nest and vice versa

e) Problems faced:

Fine Grained Job Scheduling Algorithm does not take dynamic resource characteristics into account. Can't utilize resource sufficiently.

In Artificial Fish Swarm Algorithm many parameters have impact on the final optimization result.

Ants often take time to start building good solution because it takes little iteration before pheromone trail is populated with job processor pairing.

f) Modified Algorithm:

To overcome problem an Adaptive Fine grained Job Scheduling (AFJS) algorithm is proposed

To overcome this problem and to increase the efficiency Modified Artificial Fish Swarm Algorithm(MAFSA) is introduced

To improve performance of grid system and achieve better scheduling Modified Ant Colony Optimization (MACO) algorithm is used

g) Usage:

Fine Grained Job Scheduling Algorithm uses Greedy algorithm and FCFS method.

Artificial Fish Swarm Algorithm uses Object Oriented Analytical Method

Ant Colony Optimization uses the Historic algorithm

h) Behavior:

Fine grained job can be grouped into several new jobs using constraint condition. Fine grained job are sorted using bucket sort. When the first job does not satisfy condition then Binary search and backtracking is used.

Artificial Fish Swarm Algorithm includes two parts variable and function.

Functions are: AF-prey, AF-swarm, AF-follow, AF-move.

Modified using two aspects: The leaping behavior Adaptive step length

A colony of ants begins with no solutions. Each ant constructs its own solution by making decision, using exiting problem constraints and heuristic combined with experience which is analogous to a substance called pheromone. Modified pheromone updating rule is proposed

i) Efficiency:

Fine Grained Job Scheduling Algorithm schedules coarse grained jobs and then by reducing transmission time and increasing CPU utilization, the algorithm enhance the efficiency of the system With modified AFSA and then using it to build job scheduling method it can deliver efficient performance The modified updating rule makes the ant colony algorithm to work more efficiently than the original algorithm. Thus, the Grid Scheduling problem can be easily overcome

VI. CONCLUSION

The field of Grid Computing being very vast, in this paper main aspect of scheduling is discussed. Job scheduling deals with various scheduling algorithms thereby handling many obstacles which come in the path of processing of jobs by CPU. The algorithms studied and compared are Artificial Fish-Swarm Algorithm, Fine-grained algorithm and Ant colony optimization algorithm. A comparative study of the Job Scheduling algorithms is also enclosed in this paper.

REFERENCES

[1] Junzhou Luo, Peng Ji, Xiaozhi Wang, Ye Zhu, Feng Li, Teng Ma, Xiaopeng Wang,"RESOURCE MANAGEMENT AND TASK SCHEDULING IN GRID COMPUTING" Proceedings The 8th IEEE International Conference on Computer Supported Cooperative Work in Design

[2] Lianguo Wang Yi Hong,"A MULTIAGENT ARTIFICIAL FISH SWARM ALGORITHM" Proceedings of the 7th World IEEE Conference on Intelligent Control and Automation June 25 - 27, 2008, Chongqing, China

[3] Saeed Farzi, "Efficient Job Scheduling in Grid Computing with" MODIFIED ARTIFICIAL FISH SWARM ALGORITHM", International Journal of Computer Theory and Engineering, Vol. 1, No. 1, April 2009 .

[4] M.Y.Jiang, D.F.Yuan, "Wavelet Threshold Optimization with" ARTIFICIAL FISH SWARM ALGORITHM, " in Proc. of the IEEE International Conference on Neural Networks and Brain, (ICNN&B'2005), Beijing, China, 13-15, Oct. 2005.

[5] Yeqing Liao and Quan Liu ," RESEARCH ON FINE-GRAINED JOB SCHEDULING IN GRID COMPUTING",I.J. Information Engineering and Electronic Business, 2009 .

[6] http://en.wikipedia.org/wiki/Granularity

[7] Husna Jamal Abdul Nasir, Ku Ruhana Ku-Mahamud, Aniza Mohamed Din,"LOAD BALANCING USING ENHANCED ANT ALGORITHM IN GRID COMPUTING" Second International IEEE Conference on Computational Intelligence, Modelling and Simulation

[8] Siriluck Lorpunmanee, Mohd Noor Sap, Abdul Hanan Abdullah, and Chai Chompoo-inwai," AN ANT COLONY OPTIMIZATION FOR DYNAMIC JOB SCHEDULING IN GRID ENVIRONMENT", World Academy of Science, Engineering and Technology, 2007.

[9] Mr. P.Mathiyalagan, S.Suriya and Dr.S.N.Sivanandam," MODIFIED ANT COLONY ALGORITHM FOR GRID SCHEDULING" P.Mathiyalagan et al. / (IJCSE) International Journal on Computer Science and Engineering Vol. 02, No. 02, 2010.

PUBLIC KEY EMBEDDED GRAPHIC CAPTCHA'S

ANJALI GUNWANT[1] & MANJALI GUNWANT[2]

Abstract- This paper we introduce the concept of Public key embeddedGraphicCAPTCHA's and their usage as an Anti-Phishing mechanism. By virtue of a built-in one/two-way implicit challenge mechanism and this verifiable association between the image object and the specific sub-pattern of the Public Key, they help in detecting/resisting automated or human-assisted Phishing attacks. We have presented a mutual authentication mechanism based on simple identification of an image and text within a CAPTCHA. We also merge another CAPTCHA with this that CAPTCHA is addition of Four digit numbers for providing more security The solution is based on the proposed concept of Public-Key embedded Graphic CAPTCHAs, which en-code a challenge based on a unique mapping between Image object types and bit positions of the Public Key of the website. We have also described how the proposed solution can augment the legacy Password-based authentication mechanisms and make them resistant to Man-in-the-Middle Phishing attacks. We develop these two CAPTCHA's for providing secure online banking.

I. INTRODUCTION

Captcha introduced a new genre of problems which are easy to generate but hard tosolve forcomputers.Their typical usage has been to ensure that humans are accessinga web-site instead of an automaton or stopping bots from reading/downloading contenton the Internet. The simplest one, being the trick to pull in a different human-user to solve theCAPTCHA rather than the one it was intended for or launching a Man-in-the-Middle(MitM) attack to get the human user to solve the CAPTCHA but still pick up thecontent. Though graphic object-based CAPTCHAs remain a mostly effective mechanismof ensuring human involvement, they are insufficient in thepresence MitM attacks orensuring that the intended human user is solving it. The problem of MitM attacks canbe solved (at least theoretically) by establishing a secure Transport Layer Security (TLS)-basedconnection, which builds on the strength of Public-Key cryptography. Various Phishing attacks have repeatedlytargeted this weakest linkby different schemes.After the successful installation of the Public-key certificates, which are the first and last lines of defense, not much can be really done. Whilegiving a false security impression, via the use of secure HTTP sessions, the Phisher succeeds in extracting User Login information.Further, it may also impersonate the usertowards the authentic web-site and launch a MitM attack on all data going in betweenvarious mechanisms have been proposed to counter the phishing problem, butmost of them have relied on the need for alertness and observation skills from Users..Indeed, today Phishing constitutes a primary attack by which Users becomes victim of Identity theft,compromisingnot only their Username and Password, but also the associated Identity information. Additionally, they also provide for easy provisioning and life-time manageability. However, with very low intrinsic entropy, they have also ended up easy targets for dictionary-based attacks. With this background, the goal of this paper is the design of an easy-to-use Anti-Phishing mechanism, augmenting the popular Username / Password based authentication, yet without any need for alertness or intelligence on the part of Users. In this paper, we propose a novel mechanism wherein the power of CAPTCHAs (viz. not being easily solvable by automata) is coupled with embedding unique secure channel characteristics, for e.g. a TLS-session invariant (which cannot be duplicated nor falsified) information from the underlying channel security mechanism.

Section II protocol scenario .In Section III use of Barcode reading is discussed and given. In Section IV overall application architecture is discussed. Finally, in Section V applications are discussed.

II. PROTOCOL SCENARIO

CAPTCHA encoded with partial Public Key:The Public key associated with a website is tightly coupled with the web-site's identity, as exposed within the ServerCertificate message within a TLS handshake protocol.The Public Key cannot befalsified once it has been exposed and used at the time of secure channel establishment. At the same time, it cannot be simply duplicated from the authentic website by the Phisher, as it would require the corresponding Private key for successful decryption.

Forcing a connection towards the authentic website:It is easy to incorporate this requirement within our Public KeyembeddedCAPTCHAs, by defining the authentic website to maintain the Image List and the Bit Position Map, on a per user basis. Thus, it is not possible for the Phishing website to launch an attack on the User, by generating a CAPTCHA including a sub-part of its own Public key.

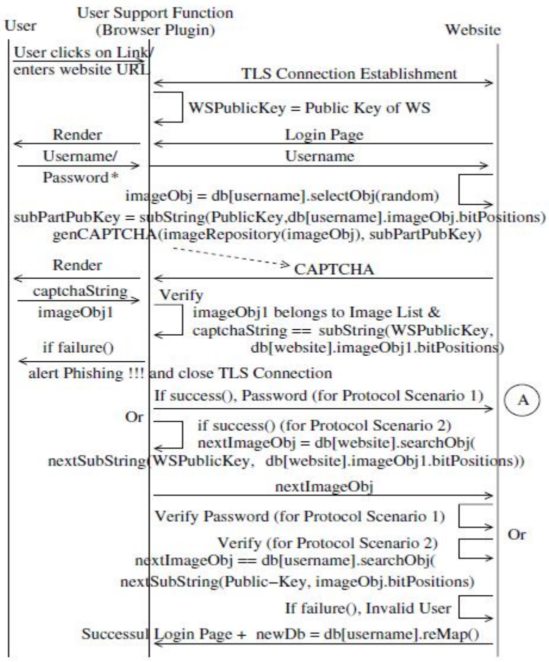

Fig. 1. Protocol Scenarios

List of the various type of Objects which can be included within the CAPTCHA. For e.g. Image List = ["car", "house", Bit Position Map = {"car" → 0-31 bits, "house" → 32-63 bits} etc. ...]

Identifying the authentic Website at the User's Browser:For knowing which graphic object types to expect within theCAPTCHA, as well as the corresponding mapping to specific Public-Key sub-parts, the User's Browser would need to store that information, indexed with the name of the Authentic Web-site. We propose that, this name be picked up from the identity which was exposed within the Server certificate, at the time of TLS session establishment. Validating the CAPTCHA is solved by the intended user:

In the Public Key-embedded Graphic CAPTCHA scheme described herein,the implicit challenge could be defined to mean "return the type of the image object,which corresponds to the xth(can be user-specific) sub-part after the sub-part (of the Public Key), which was encoded within the CAPTCHA".This would require the human-assisted Phisher to guess the Image List for the User as well as the Bit Position Map.However, an authentic User's system cans easily lookup the Image List and Bit Position Map to respond to the challenge. This, forces the Phisher to deliver the Authentic Web-site's CAPTCHA towards the User's system.

III. PROPOSED SYSTEM

Registration Phase: The aim of this phase is to allow a user and a server to negotiate a shared secret toauthenticate succeeding logins for this user.

Login Phase: Used to login an application by entering the user ID, then selecting correct name of `image from the given list and enter text which is displayed on the image.

Recovery Phase: Recovering from an Identity Attack: If it does happen that the User is attacked (fore.g. due to access from a Public Computer), then they need a backup mechanism torecover the right to their username. To enable this, the USF within the Users trustedsystem (for e.g. personal computer) may collect and store the various CAPTCHA-basedchallenges that it receives over a period of time.

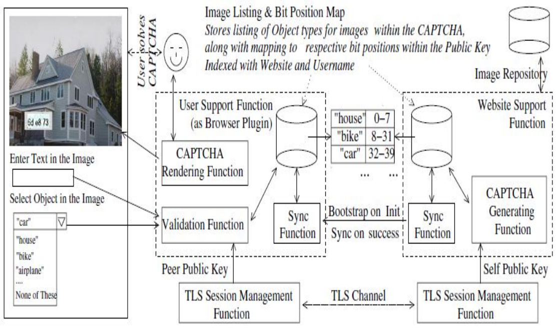

Fig.2.Functional Architecture for generation and usage of Public Key embedded Graphic CAPTCHAs, with User Interface screenshot

IV. APPLICATION

This application will be useful for the many Online Applications to provide more security with two level Captcha's (i.e. Graphics captcha and Arithmetic captcha).And here we are providing this security for secured online Authentication using CPTCHA.

REFERENCES

[1] Takumi Yamamoto, Tokuichiro Suzuki, MasakatsuNishigaki1. A Proposal of Four-panel cartoon CAPTCHA: The Concept.Network-Based Information System(NBiS),2010.

[2] L. von Ahn, M. Blum, N. Hopper, and J. Langford. Captcha: Using hard ai problems for security. In Proceedings of Eurocrypt, pages 294–311, 2003.

[3] Chandavale, A.A. Algorithm for secured online Authentication using CPTCHA.Emerging trends in Engineering and Technology(ICETET),2010.

[4] Amir Herzberg and Ahmad Gbara. Security and identification indicators for browsers against spoofing and phishing attacks. Cryptology ePrint Archive, Report 2004/155, 2004.

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LINUX SETTING MANAGER

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Abstract- In the present system, users have to manually select the specific application or configuration utility for different settings like network, firewall, kernel modules, etc. However, in many Linux distributions, even this functionality is lacking a GUI front end. Hence, it becomes cumbersome for novice users to set up their settings through the terminal. Remembering and entering command-line options is a tedious task and many times errors can occur due to simple mistakes in commands or due to user's misunderstanding of the extensive parameters available. Even for experienced users, while setting up a new machine or changing some configuration settings quickly and easily, there is a need for a better and more user-friendly GUI alternative to get the job done. Considering the above mentioned drawbacks of the present system, it is obvious that a centralized program with GUI will be helpful to all Linux users, novice and expert alike. A central place for all major settings that users perform will not only make configuration an easy task but will be helpful in saving time and other errors that may arise while doing it manually through command-line.

I. INTRODUCTION

The existing system does not support making profiles for different settings in the system. It requires certain settings to be done. Every time a setting is changed, the previous setting is lost. The user has to remember commands for changing each setting. There is no way of saving which loadable kernel modules are in use. If the user wants to load a new module, he has to do it every time the machine is started. We propose a set of adaptive strategies that help in taking better GUI based model. The GUI enabled application helps user to make his own settings easily. The GUI will enable the user to change many settings such as network configuration, firewall configuration and Loadable Kernel Modules (LKM). Some of the features can be enabled or disabled depending on the profile. These different features can be saved and can be loaded easily. There will be user preferences given which will help user to make changes in system easily. Also, these profiles can be saved to a file which can then be sent to anyone anywhere to get the same settings at the other PC.
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II. RELATED WORK

In Linux GUI is developed using qt designer that is present. [1]Networking related files and commands used are referred from red hat [2] . Linux had grown to be as stable and as reliable as a very powerful operating system. Linux kernel is 'copylefted' software, patented under the GNU GPL, and thus, nobody actually owns it. But more significantly, Linux is sheltered by the Open Source community.[3]

III. PROPOSED SOLUTION

Considering the above mentioned drawbacks of the present system, it is obvious that a centralized program with GUI will be helpful to all Linux users, novice and expert alike. A central place for all major settings that users perform will not only make configuration an easy task but will be helpful in saving time and other errors that may arise while doing it manually through command-line.

Now a day, it has become increasingly important to make software to save users' time and efforts doing monotonous and unnecessarily time consuming tasks. This helps users concentrate on the real work at hand instead of spending time and effort setting up the system. In addition to this, a portable one-click solution that we propose will make it very easy for all users to have a consistent system experience even if they are using different machines or networks. Hence, it helps users work just as efficiently as they would at their homes or workplaces using their own personal computers. There will no longer be a need to remember one's personal settings, work settings, or the array of commands that accompany them. Linux has been avoided by many a novice users because of its lack of user friendly UI. Our software helps to tackle this problem in the configuration department, which is very commonly used by many users.

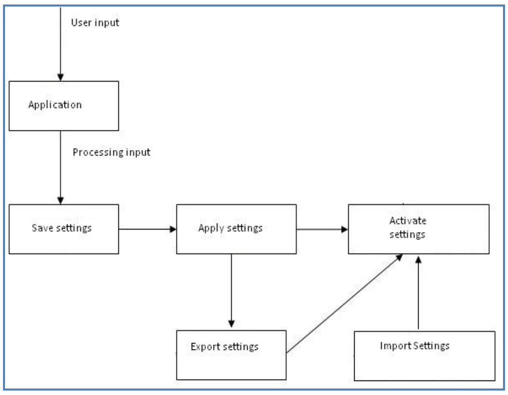

Figure1: This explains the system design of LSM.

IV. FUNCTIONAL SPECIFICATION

To create Linux Settings Manager, a GUI based software which will provide a central settings management location for Linux users. It will host the most commonly required configuration options for immediate or later use. Profiles will allow multiple users as well as offer portability to the users. The system is aimed at all Linux users as an alternative to command-line arguments required to achieve the same goal. It will provide the following functions and features:

1) GUI for network, firewall, kernel modules settings
2) Ability to create different profiles for multiple users
3) Ability to store settings of each user in personal settings file
4) Easy loading of settings from settings files.
5) Instant application of all settings selected
6) Option to reset settings if needed
7) Support for multiple network adapters

V. CONCLUSION

This paper presents a practical implementation of a Linux Setting Manager.

ACKNOWLEDGMENT

First and foremost, I would like to thank my guide, Prof. Mrs S. M. Murge, for her guidance and support. I will forever remain grateful for the constant support and guidance extended by guide, in making this report. Through our many discussions, she helped us to form and solidify ideas. The invaluable discussions I had with her, the penetrating questions she has put to me and the constant motivation, has all led to the development of this project.

REFERENCES

1. "property setting manager for objects and controls of a gui software development", daniel j. Meziek, united state patent
2. "red hat linux networking", pete nesbitt
3. The linux project and microsoft vs. Linux chapters from management & virtual decentralized networks: the linux project.
4. Improving reliability with restartable file systems, swaminathan sundararaman, sriram subramanian, abhishek rajimwale, andrea c. Arpaci-dusseau, remzi h. Arpaci-dusseau, michael m. Swift, 2010

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PROCEDURAL MUSIC GENERATION FOR GAMES USING EXPERIENCE DRIVEN APPROACH

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Abstract- As video games have grown from crude and simple circuit-based artefacts to a multibillion dollar worldwide industry video-game music has become increasingly adaptive. Composers have had to use new techniques to avoid the traditional, event-based approach where music is composed mostly of looped audio tracks, which can lead to music that is too repeti tive. In addition, these cannot scale well in the design of today's games, which have become increasingly complex and nonlinear in narrative. This paper outlines the use of experience-driven procedural music generation, to outline possible ways forward in the dynamic generation of music and audio according to user gameplay metrics.

Index Terms— *Adaptive algorithm, genetic algorithms.*

I. INTRODUCTION

Today, game music reacts adaptively to player experience, foreshadowing upcoming events in gameplay, and adding impact to major events. Instead of composing music to specific game cues, composers are now asked to compose not only different versions of the same cue (varying in intensity or instrumentation), but also versions that adapt to changes in player style.

In recent games, which usually contain very large open worlds with open-ended narratives, it is often impossible to know what exact events will occur in the game during the length of a particular audio segment. If a segment is simply looped, it leads to tedium for th e player, a problem acknowledged by Tomas Dvorak, composer of the soundtrack to Machinarium [1]: "Soundtrack music has to be more abstract to give space for the image and also to not be annoying if it repeats" [2]. As on e of the mo st important roles of music in game design is to immerse the player in the gameplay through emotional induction, avoiding tedium is paramount. This is particularly true in mobile applications, which live and die by game play length (time between launch and app termination), where emotional engagement is therefore crucial, as mobile operating systems typically kill the application right away when a "home" button is used.

Procedural content generation (PCG) refers to the programmatic (algorithmic) generation of game content. So far, it has been used to produce dynamic, as opposed to preco mputed , content such as light maps and levels. As a set of techniques, it of-fers great advantages toward creating music that adapts more granularly to player experience, avoiding needless repetition and providing an evolving, more emotionally intelligent sound track. Repetition itself, in terms of composed music for games, should, for the con text of this paper, be understood to be inherently counterproductive. Indeed, repetition in music is a known device, and has been proven to have an effect on liking [3]. Studies locating repetition within an overall theory of musical syntax to better understand its role in popular music, for example, have usefully outlined its success as a mechanism of musical production [4]. However, there is often a direct disconnect between the near nonlinearity of recent games, which can contain open worlds and narratives, and the use of repetitive music: often, users find the tail ends of composed pieces of music simply "restarting," having run out of timed gameplay.

This has prompted new games to have an increased number of cues, for composers to work to. But as Collins examines, "composers now commonly re-use cues in other areas of a game, to reduce the amount of unique cues needed, but without creating a repetitive sounding score" [5]. We believe that some of the techniques outlined in this paper, while obviously not meant to replace the work of a composer, or to label all repetition erroneous or facile, could help in bridging the gap between the increasingly large scale of games and their music, the composer's dilemma, and what Collins terms "listener fatigue" [5, p. 269]. Procedural music can be defined as "composition that evolves in real time according to a specific set of rules or control logics" [6,p.13].Morespecifically, procedural generation of audio can mean that audio events are stored as code and unpacked when triggered to synthesize audio in real time, requiring only text-based storage as opposed to sampled audio loops, offering significant memory and storage savings.

Notwithstanding this advantage, PCG is not yet common in game design. Primarily, this is because the complex control logics th at PCG demands are at odds with traditional budget allocation for music, which must compete with graphics. Hiring traditional composers is simply cheaper than hiring audio programmers. Also, procedural audio tends to be central processing unit (CPU)-intensive and difficult to tie logistically to meaningful game elements [6, p. 12].

However, new PCG techniques beyond the field of audio design and music composition are emerging, modeling player experience in order to create adaptive content [7]. While experience-driven PCG (EDPCG) is no w primarily used to gen cerate terrain, maps, levels, and weapons, "there is room for approaches other than those that have already been tried; both the type of content generated and the algorithmic approach to generating it may change in the future" [7, p. 13].

In the context of PCG research, content is most often generated as a result of a stochastic search process, such as an evolutionary algorithm, which will use a fitness function to evaluate whether created content is appropriate. This is done over many generations, where candidate content is ranked, some discarded, some mutated, and further generations created. The combination of procedural audio generation and experience modeling could provide ways forward in the design and composition of both diegetic sound and music in large, virtual environments such as open-world games, where traditional composition and sampled audio techniques would not scale. At the same time, it would provide more engaging player experiences in traditional linear narratives, as well as other environments currently in announced by game design, such as therapeutic systems, on-demand and in- ight entertainment systems, and web 2.0 services.

This paper investigates whether recent findings in search-based PCG (SBPCG) [8] and EDPCG could be applied to sound design and music compo sition. SBPCG refers to a special case o f the generate-and-test approach to PCG, whereby the test function does not just accept or reject candidate content, but grades it according to a fitness function [8,p. 4], and EDPCG expects to give this search-based approach a model of the user experience that it can use to "generate content that optimizes the experience for the player" [7, p. 3]. Both can be deployed in order to help avoid tedium in large open-world and open-ended game environments, as well as game-in uenced aspects of industrial and medical design. We will primarily focus on biologically inspired computational intelligence, embeddable synthesis, and player experience modeling to provide new pathways for music composition in open narrative media.

A. The Origins—Algorithmic Composition

From Markov models, generative grammars, transition networks, evolutionary algorithms, and chaos theory, to agent, neural, and cellular automata-based systems, algorithmic composition has a long history, which is now well documented [9]. Algorithmic composition's relationship to other environments, such as game design, is more recent. Entire generative models have been developed for the automated composition of music drawing on theories of emotion, perception, and cognition [10], and models from computational intelligence and stochastic computation in general [11]. Much work has been done in the area of mood tagging and effect for adaptive music, almost always using western-notational representations of music through the Musical Instrument Digital Interface (MIDI) standard to compose segments of music that match particular "moods" [12]. However, while there is previou s work in wholly synthesized, generated audio (as opposed to MIDI) in the algorithmic composition domain [13], little work has been done on adaptive game audio, synthesized in real time, that uses player experience modeling to drive adaptive content creation.

B. Precedents and Future of PCG in Game Development

Early game developers attempted to provide unique game play based on personal player style [14, p. 12], despite severely limited storage space [15]. This adaptive process was mirrored by music composers, who used repetition, transposition, and other techniques to offer variation to the player. Algorithmic sound design, which generates audio and music based on rules and behaviors, has been used since the beginning of games development. Usually combining sampled audio with synthesis, it allows for exible layering and real-time effects instead of just a stereo mixdown of precomposed diegetic audio or music. Reactive ambiences, for example, can be made of collections of sampled audio loops; a country ambience made of bird so ng, insects, and wind might die down in reaction to a gunshot fired by a player or A Identity.

As early as 1987, Iwai's Otocky [16] used the shooting controls in the game to generate different harmonic palettes in the background music, thus allowing the player to affect the game's music through gameplay itself, a concept he later developed in SimTunes , a predecessor to many of today's musical video games. Some LucasArts games, such as Monkey Island 2: LeChuck's Revenge [17], use iMuse, a software system that allows musical changes to occur at very small intervals, approximately every four measures, between a large number of branching musical segments [15], depending on the player's in-game exploration choices and nonlinear dialog selections. The music–player interaction afforded by iMuse has continued in many contemporary games, such as Tron 2.0 (Monolith Productions, 2003), XIII (Ubisoft Paris, 2003), and Heavy Rain (Cage, 2010), to offer players direct structural control over the cinematic soundtrack without giving them explicit clues as to how their choices affect this music.

Very recent developments in embeddable audio engines such as libpd [18], an adaptation of the Puredata (PD) data ow language as a signal processing library, mean that it is now possible to use PD for the creation of procedural music and game audio at an industrial level, independent of game engine or programming language. Spore (Electronic Arts, 2008) is o ne of the first examples to implement this in the core design process, with Electronic Arts and Brian Eno creating their own version of PD (EApd) to embed within th eir core engine. Kent Jolly and Aaron McLeran, the audio programmers in Spore 's team who worked with Eno, described creating adaptive/procedural music as "a different way to compose," where you are actually "composing in probabilities" [19] using game events to trigger musical variations.

In other spheres, sound generation is starting to play a key role in the gameplay itself, in games such as Papa Sangre (Something Else Studio, 2010), which use binaural real-time sound to give its users a sense of a 3-D sound world. However, as Collins points out, most algorithms used for music control are "transformational, rather than truly generative," due to the "difficulties in composing effective procedural music for games" [6, p. 8].

Namely, this is because procedural music and sound has to be bound by strict control logics in order to function adequately within the constraints of game functions, such as action anticipation, leitmotif (within and across games), reward signals, and emotional induction.

These control logics must enable the music to ad apt quickly to player input and in-game parameters, as well as be able to change adaptively when situated in long gameplay. As most modern games offer 40–60 h of scripted scenarios [6, p. 6], and players can spend hours in particularly difficult parts of a game, listening to music that is not adaptive can become tiring and, crucially, boring. Designers and composers have reacted by incorporating timing cues that will fade music out instead of looping endlessly when in long periods of gameplay in the same scenario, or in Spore's case, the "density of the instrumentation in the procedural music is reduced over time." Procedural music may thus "offer some interesting possibilities that may solve some of these complications of composing for games" [6, p. 7]. When algorithms used in procedural music are not just trans formational, and are instead generative (according to the terms discussed in [20]), they fit abstract game narratives better than set narratives with traditional plot points. More recently, both the Creatures series and Spore, relying heavily on procedural generation for the creation of their game world, have opted for more organic approaches, using very short samples of recorded music (note units) to generate the soundtrack in real time ac- cording to in-game parameters such as mood, environment, and actions. Even here, however, the algorithmic strategy is mostly transformational and simply stochastic (relying on rule-based grammars), as opposed to purely generative.

An approach to generative algorithms in the procedural generation of audio and music that would not only take into account in gamerulesystems, but also search-based, experience-driven parameters, would provide both better variety of transformational potential, and a more relevant emotional connection to the player. We chose to take inspiration from the EDPCG frame work outlined by Pedersen [21], and closely modeled player behavior in order to derive musical rules and sets of variables we could then use within our generative algorithms, which procedurally generate MIDI music in real time. Some of the musical structure is generated by a simple iterative process, and some generated by classical genetic algorithms. Below, we outline this process.

C. Frustration, Challenge, and Fun

Following the EDPCG framework, we encoded frustration challenge, and fun as functions of the metrics of the player's gameplay, as contained within the Mario AI Championship engine [22], with some modifications. Mario AI is a modified version of Markus Persson's In finite Mario Bros, which is a public domain clone of Nintendo's Super Mario Bros, and was created for the Mario AI Competition. The focus of the competition was "on developing controllers that could play a version of Super Mario Bros as well as possible" using computational intelligence[23]. We have adapted the Java source code of the Level Generation Track of this competition in order to generate adaptive music for Mario AI following the EDPCG model.

Table I
Values Chosen as Multipliers

Multiplier	Frustration	Challenge	Fun
Shells kicked	0	0	1
Running time	0	0	1
Coins collected	0	0	1
Coin blocks destroyed	0	-1	1
Monsters killed	0	0	1
Fell into gap	1	1	0
Time ducked	0	1	0
Power blocks destroyed	0	-1	0
Alive time	-1	-1	0
Time standing still	1	0	0

As our purpose was to generate adaptive mu sic that would take EDPCG into account, we decided to evaluate frustration, challenge, and fun in real time in order to implement a musical engine capable of reacting to the player's mood as it changed du ring gameplay. First, we modified the Data Recorder class within the Mario AI engine in order to derive information about the timing of events: every metric was transformed from a scalar value to an array of values, representing the number of events regarding that particular metric that occurred during a particular times pan. For example, the first scalar of array coins Collected contains the number of coins the player collected during the first second of the game (timespan 1s). Pedersen [21] showed that each of the frustration, challenge, and fun functions has certain weighted correlations with each of the gameplay metrics.

Table I shows the values that were chosen as multipliers. All the values are normalized on the time span to take into account, i.e., shells kicked: the number of shells kicked per second. Values with a positive correlation with the metric taken into account have a positive multiplier (i.e.,shells kicked and fun);values with a negative correlation have a negative multiplier(i.e.,tiplier equal to zero (i.e., alive time and challenge); values not correlated have a mul-running time and challenge).

D. Ma pping of Frustration, Challenge, and Fun to Excitement

To keep the musical generation and its rules as simple and obvious as possible, and to offer the player discernible musical changes according to his mood, we decided to map frustration, challenge, and fun to a single metric—target excitement expressing the mood we try to induce in the player through the music. In our context, we define "target excitement" following this process: we want to excite and encourage the player if: 1) he is not having a frustrating game (the contrast between the gaming experience and the music would be disturbing); 2) he is having fun; and 3) he is not challenged too far (to avoid rising frustration). In this sense, fun is the most important metric in the equation. All our variables (frustration, fun, and challenge) are EDPCG metrics, implemented following its core literature.

To implement target excitement, we assigned weights to each metric target excitement fun challenge frustration Fun was then positively correlated to excitement as its most important indicator. Frustration and challenge should be negatively related to excitement: the more difficult the game is for the player, the more calming the music should be. We decided to use the pentatonic mode as the calming target. The pentatonic and minor scale is used to achieve a calming effect on listeners because it is the most common scale worldwide, primarily because it lacks any dissonant intervals between its pitches. This makes the pentatonic scale unique in that any of its pitch members can be combined without harmonic clashes, and therefore offer the listener no acoustic challenges. We focused on brightness and major modes to react to excitement, in order to celebrate ow by rewarding the user with obviously joyous-sounding music. By traveling from exciting, bright, major-mode musical structure to pentatonic, soft-filtered structures, we hope to both reduce the frustration of playing a difficult game, and contribute to a state of ow when challenge and skill are matched.

E. Generative Music Engine

Music is generated at runtime using frustration, challenge, and fun as generative parameters. The generative engine has been conceived to be as simple as possible, yet able to produce simple tonal music. It is composed of the following parts, both of which follow a classic genetic algorithm (GA) structure of (Fitness)Selection–¿Crossover–¿Mutation–Offspring:

• harmonic sequence generator;
• period builder for the melodic line.

Every generator (chord sequence, melody) breeds new candidates every time the current item is ready (i.e., the last chord of the current sequence has ended). The fitness function is then used to select the winner from the candidates. Both the harmonic sequence generator and the period builder work in real time, creating musical structures reacting to the excitement metric. The music is generated as a stream of MIDI events.

The generators react in real time to the excitement metric with the following rules:

• Beats per minute: 120 beat/min for minimum excitement(relaxed) and 135 beat/min for full excitement;
• Scale: pentatonic scale is preferred for minimum excitement, and full major scale is preferred for excitement;
• Novelty: repetition of already heard material is preferred for minimum excitement, and introduction of new material is preferred for excitement;
• Sparseness: phrases with sparse notes are preferred for minimum excitement, and dense phrases are preferred for excitement;
• Resonant filter: low resonant filter is preferred for minimum excitement, and high resonant filter is preferred for excitement.

1) Harmonic Sequence Generator: The harmonic sequence generator works on a basic idea of harmony: chords. It is responsible for deciding the next group of chords with respect to a history of all already played chords. A chord is expressed as a main pitch over a particular scale. For example, the C major chord in the C major key is a couple (main pitch, scale) where main pitch (where 36 is the central C note in MIDI), and scale , the C major scale in

MIDI notes; the fifth grade of the C major key would then have the same scale as the previous example but 43 as the main pitch (a G note). We give the engine the chord sequences (harmonic functions) that we want to be used by the g enerator, for example, For simplicity, we only insert chord sequences in a single key (C major).

At runtime, every time a new chord sequence is needed, all existing chord sequences are evaluated by the fitness function:
The chords contained in the sequence are matched against the history of already played chords. An index of novelty is then assigned to each sequence. Sequence has a high historic novelty if it has not been heard before and a high inner novelty if it contains many different chords.

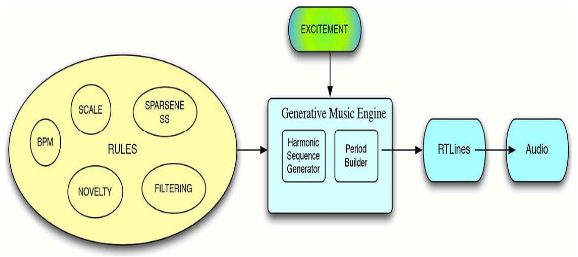

Fig. 1. The Period and Harmonic generators at work.

Sequence with the minimum is chosen.
2) Period Builder: As well as chords, phrases are another basic compositional notion we use in our engine. For our purposes, phrase is a sequence of single MIDI notes.

A small set of phrases (, , ,and , called seeds) is defined at compile time, which are hardcoded. A large number of variations of each seed is then generated (, etc.). A variation is a small change in a phrase where a random note is either deleted, added, or modified, and we use this as a basic transformative mechanism in crossover.

Aperiod is a sequence of p hrases:
As it of phrases is defined as a compile time and a large number of variations is generated. A variation is the substitution of a phrase with one of its variations (is a variation of), which we use as the basic process of mutation. Similarly to what is done with the harmonic sequence generator, every time a new period is needed, all the existing periods are evaluated by fitness functio ns: the period is matched against the history of already played periods, using the historic novelty formula definedinSectionI-E1historic novelty (10) Also, the density of a period is defined as the average value of the density of the phrases. The density of a phrase is defined as the number of notes in the phrase divided by 16 (the maximum number of notes a phrase can hold in its genotype) density density (11) where is the number of phrases in the period and density excitement histo ric n ovelty density (12) The period with the minimum is chosen.

F. JMusic RT Lines
Throughout the definition of genotypic material such as phrases and chords, we use JMusic [24], a Java environment for music composition designed to assist the compositional process by providing an o pen but partially structured environment for musical exploration. JMusic was chosen because it fit with Mario AI's Java environment, and because it already contained musical notions that we could use to model variables and game metrics from player experience into audible events, contained in Java classes such as Note and Phrase. While JMusic is primarily designed to create music of ine, a few classes for real-time generation of music, notably RTLine, can be used to generate music on the y. Our Harmonic and Period generator helps to build such a stream using an RT Composition, a class that amalgamates the different generative streams within our engine, and composes RTLines using a synthesized instrument, Saw LPF In stRT2, which is then used to transform the note-by-note information from our stream into synthesized audio.
Fig. 1 outlines the audio engine's process.

II. TESTING

We carried out a small perceptual study that embedded the Mario AI Applet in a Java-Web Start web app running Node.JS and MongoDB. We modified the Applet to send JavaScript Object Notation (JSON) data directly from our embedded gameplay metrics. The study ran for two rounds each game, and two games per tester. After the first game, a small form was presented to the user to ask pertinent questions (Are you a musician? Have you played Mario before? Did you enjoy this game?). After the second game, only enjoyment was measured through the questionnaire. Question results and metrics were then bundled into JSON packages and sent to our server.

In all, we ran it through 31 testers, eight of which were non musicians, 17 played an instrument but were not musicians, and six were actual musicians. Our results, from a small userbase, were inconclusive, although in teresting. We would like to expand this next, with adequate time and social advertising.

A. Test Data
All our metrics were expressed in a oating point scale from 0to1.In all two-round games, the frustration during the first round was generally higher than during the second round: 90th percentile in the first round was 0.48 (average value 0.11 with 0.29 standard deviation), while in the second round, it was 0 (average value 0.00 with 0.01 standard deviation). Frustration was 0 on 77% of the time in the first round and 0 on 93% of the time in the second; this

indicates that the second round was almost always easier for all the players while the first round was sometimes frustrating. This was to be expected, as less experienced users get used to controls and game dynamics during their first round. We built a randomization system that presented either a game round with generative music, or a pre-composed MIDI track alternatively, each time a test was requested. Of all tests, the first round was generative 0.61 of the time. Contrary to our expectations, average frustration during the generative round was equal to the MIDI round, with 0.08 (with 0.25 standard deviation) for generative and 0.03 (with 0.15 standard deviation) for MIDI rounds. Generative rounds were prevalent during the first round, which we think probably biased bad results for generative rounds. However, the frustration average during generative rounds was consistently lower than the frustration average during first rounds overall, which could be taken to mean that generative music helped in lowering player stress during the first round of play; however, without adequate measurement systems such as galvanic skin response and heart rate, and an adequate clinical protocol with physically present users, as opposed to remote ones, we cannot offer proof of this.1) Enjoyment and Fun—Similar Results: We found that fun levels in our metrics looked approximately the same during gameplay (average fun during generative round: 0.61 with0.29 standard deviation ; average fun during MIDI round: 0.64 with 0.29 standard deviation). Enjoyment also looked the same during generative and MIDI rounds (average enjoyment during generative rounds was 0.66 with 0.35 standard deviation and average enjoyment during MIDI rounds was 0.67 with 0.33 standard deviation). Users reported the same level of enjoyment 77% of the time, which gave us inconclusive results as to whether generative music, which may have helped lower stress during first rounds, contributed at all to overall levels of enjoyment. Nonetheless, such a high percentage of users selecting the same level of enjoyment could indicate a poor test design: users may have given the same answer because we offered only three choices ("I enjoyed this round," "I almost enjoyed this round," and "I didn't enjoy this round") leading to users giving the same answer for both rounds. When the users reported a different level of enjoyment in the two rounds, the generative round was preferred 57% of the time. More testing and better test design is required to understand if the slightly higher number of preferred generative rounds is statistically relevant. B. Future Testing and Work While the user testing group was too small to give us enough data to successfully prove generative music helped ameplay, we hope to test with larger groups and using a better test design later on. We are encouraged that frustration averages seemed consistently lower during generative rounds, and we hope that on much larger groups, we could gather conclusive data to illustrate this effect. We hop e to leave the experiment running at its base URL, and to point to it from ongoing Face book advert campaigns, over a period of a few months.

III. C ONCLUSION

While algorithmic composition and procedural music both have a long history as academic disciplines and within the game industry, player behavior modeling, and in particular, adaptive audio engines that listen to such a model are still rare. While games like Sp ore go a long way to ward embedding normally academic music and synthesis engines like Pure data into studio produced games, the state of the art still relies on engines such as iMuse, technology from 1991, for seamless musical component building that adapts to player decisions, trying to connect pieces of music as the player moves from area to area, using the idea of seamless musical "bridges" to connect major pieces of music. This paper proposes that paradigms such as EDPCG, and ideas from computational intelligence in use in player experience modeling, could transfer into adaptive musical composition for games. It proposes the use of tried-and-tested evolutionary algorithms subsuming the idea of EDPCG as o ne example of using computational intelligence to achieve this. We think that such an approach could help build adaptive audio engines to supersede tools such as iMuse, pointing the way forward for music in games that is not just linear and scene reactive, without necessarily demanding large investment. While libraries like JMusic are practical for the purposes of engaging with a competition-driven environment such as Mario AI, embedded composition environments such as libpd can now be used and embedded in most major programming languages. This makes them accessible to popular game engines such as Unity and indeed iOS/Android and HTML5 environments, which puts decades of research into algorithmic composition and real-time audio synthesis at the fingertips of not just AAA studios, but also amateur and indie developers, from whom so often come gameplay ideas that change the field.

REFERENCES

[1] Machinarium [Online]. Available: http://machinarium.net/

[2] Game-OST [Online]. Available: http://game-ost.com/articles.php?id=76action=view

[3] D. J. Hargreaves, "The effects of repetition o n liking for music," J.Res. Music Edu. , vol. 32, no. 1, pp. 35–47, 1984.

[4] R. Middleton, "'Play it again Sam': Some notes on the productivity of repetition in popular music," Popular Music , vol. 3, pp. 235–270, 1983.

[5] K. Collins, "An introduction to the participatory and no n-linear aspects of video games audio," in Essays on Soundand Vision ,S.Hawkinsand J. Richardson, Eds. Helsinki, Sweden: Helsinki Univ. Press, 2007, pp. 263–298.

[6] K. Collins, "An introduction to procedural music in video games," Contemporary Music Rev. , vol. 28, no. 1, pp. 5–15, 2009.

[7] G. N. Yannakakis and J. Togelius, "Experience-driven procedural content generation," IEEE Trans. Affective Comput. , vol. 2, no. 3, pp. 147–161, Jul.–Sep. 2011.

[8] J. Togelius, G. Yannakakis, K. Stanley, and C. Browne, "Search-based procedural content generation," in Applications of Evolutionary Computation , ser. Lecture Notes in Computer Science. Berlin, Germany: Springer-Verlag, 2010, vol. 6024, pp. 141–150.

[9] G. Nierhaus , Algorithmic Composition: Paradigms of Automated Music Generation . New York: Springer-Verlag, 2009.

[10] D. Birchfield, "Generative model for the creation of musical emotion, meaning, and form," in Proc. ACM SIGMM Workshop Exp. Telepresence, 2003, pp. 99–104 [Online]. Available: http://doi.acm.org/10. 1145/982484.982504

[11] W. Hsu, "Strategies for managing timbre and interaction in automatic improvisation systems," Leonardo Music J. , vol. 20, pp. 33–39, 2010.

ANALYSIS OF HUMAN ATTENTIONAL NETWORKS

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Abstract- In this paper we outline a conception of attentional networks arising from imaging studies as connections between activated brain areas carrying out localized mental operations. We consider both the areas of functional activation (nodes) and the structural (DTI) and functional connections (DCM) between them in real time (EEG, frequency analysis) as important tools in analyzing the network. The efficiency of network function involves the time course of activation of nodes and their connectivity to other areas of the network. We outline landmarks in the development of brain networks underlying executive attention from infancy and childhood. We use individual differences in network efficiency to examine genetic alleles that are related to performance. We consider how animal studies might be used to determine the genes that in uence network development. Finally we indicate how training may aid in enhancing attentional networks. Our goal is to show the wide range of methods that can be used to suggest and analyze models of network function in the study of attention.

Keywords- *Alerting; DCM (Dynamic causal model for connectivity analysis); DTI (Diffusion tensor imaging); EEG frequency (electroencephalographic); fMRI (functional magnetic resonance imaging); Executive attention; Orienting; Self-regulation*

The goal of this paper is to inform people interested in neural network models about efforts to analyze the networks of neural areas revealed in imaging studies and to understand how genes and experience shape their development. To do this we first discuss attention networks as defined by anatomical areas active in functional imaging studies, and then examine physical and functional connections between these areas. Next we consider how these networks develop during infancy and early childhood and finally what is known about how genes and experience shape the network.

I. NETWORKS OF ATTENTION

What do we mean by an intentional network? In cellular physiology the idea of a network involves identified neurons that connect to one another by synapses and through other means of communication (Bullock et al., 2005). Connectionist models, inspired by neural networks, have considered units at particular levels that in uence each other by direct or reciprocal connections (O'Reilly & Munakata, 2000). Imaging of human task performance has identified another level of network function, which is clearly related to both the models and the underlying cellular structure by showing that a number of quite separate brain areas must be orchestrated in even the simplest task. Each of these areas may be performing a different computation, which taken together allow performance of the task. We regard the set of activations and their connections as the network that underlies task performance. It is often believed that attention is a general property of the whole brain, but neuro imaging studies have shown specific networks of neural areas are involved in functions related toattention (see Fig. 1). Attentional networks are special in that their primary purpose is to in uence the operation of other brain networks. As illustrated in Fig. 1 three attentional functions for which brain networks have been imaged are: alerting which is involved in acquiring and maintaining the alert state; orienting to sensory stimuli and executive control involved in the resolution of con ict between neural systems and regulating thoughts and feelings (Fan, McCandliss, Fossella, Flombaum,

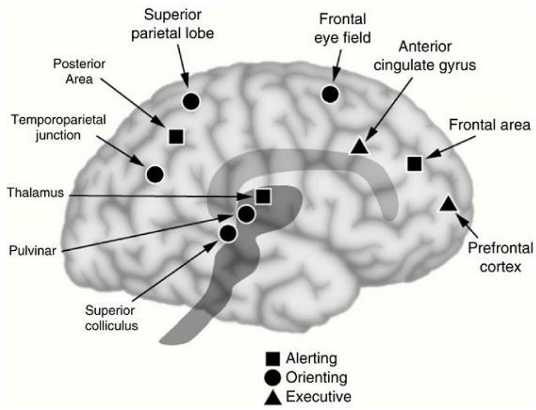

Fig. 1. Thisfigure illustratescortical areas involved in three attention networks.

The alerting network (squares) includes thalamic and cortical sites related to the brain's nor epinephrine system. The orienting network (circles) is cantered on parietal sites (discussed below) and the executive network (triangles) includes the anterior cingulated and other frontal areas. & Posner, 2005). Although the sites at which attention in uence can be demonstrated involve most any brain area including primary sensory, limbic and motor cortex, as shown in Fig. 1 the sources of these activations are much more limited. Orienting to sensory events has been the more studied of these networks both with imaging (Corbetta & Shulman, 2002 and cellular (Reynolds, 2004) methods. The convergence on the set of brain areas serving as the source of the amplification of sensory signals has been impressive (see Hillyard, Di Russo, and Martinez (2004) , for a recent review). It is widely agreed that the frontal eye fields work in conjunction with superior and inferior parietal areas as the cortical nodes of the orienting network. In addition, studies have implicated some subcortical areas including the pulvinar of the thalamus and the superior colliculus. Most of the studies of this network have involved visual stimuli, but the sources of the attention in uences in orienting to other modalities are much the same. Of course the site of amplification of the sensory message is quite different for each of the modalities.

Evidence to date suggests that both maintained alertness during task performance (tonic) and phasic changes induced by a warning signal involve a sub cortical structure, the locus ceruleans that is the source of the brain's nor epinephrine. A great deal of evidence (summarized in Posner and Petersen (1990) indicates that the tonic state depends upon an intact right cerebral hemisphere. Lesions in this hemisphere can produce profound difficulty in responding to unexpected targets. Warning signals, however, may have their in uence more strongly on the left cerebral hemisphere (Coull, Frith, Buchel, & Nobre, 2000; Fan et al. , 2005). This distinction may react a more general division between the hemispheres where rapidly acting events are left lateralized while more slowly changing states involve right hemisphere activity. Tasks that involve con ict between stimulus dimensions competing for control of the output often provide activation in the anterior cingulated gyros and lateral prefrontal areas.

It is thought that the con ict, induced by a stimulus, is representative of situations where different neural networks compete for control of consciousness or output. Because of this we have termed this the executive attention network because it regulates the activity in other brain networks involved in thought and emotion (Crottaz-Herbtte & Mennon,2006 ; Etkin, Egner, Peraza, Kandel, & Hirsch, 2006). This network shows a strong development in childhood and its maturation is related to what in developmental psychology has been called self regulation.

Individual differences are invariably found in cognitive tasks involving attention. The Attention Network Test (ANT) was developed to examine individual differences in the efficiency of the brain networks of alerting, orienting and executive attention discussed above (Fan, McCandliss, Sommer, Raz, & Posner, 2002 ; Rueda , Fan , et al. , 2004). The ANT uses differences in reaction time (RT) between conditions to measure the efficiency of each network. Each trial begins with a cue (or a blank interval, in the no-cue condition) that informs the participant either that a target will be occurring soon, or where it will occur or both. The target always occurs either above or below fixation and consists of a central arrow, surrounded by anking arrows that can either point in the same direction (congruent) or in the opposite direction (incongruent). Subtracting RTs for congruent from incongruent target trials provides a measure of convict resolution and assesses the efficiency of the executive attention network. Subtracting RTs obtained in the double-cue condition from RT in the no-cue condition gives a measure of alerting due to the presence of a warning signal. Subtracting RTs to targets at the cued location (spatial cue condition) from trials using a central cue gives a measure of orienting, since the spatial cue, but not the central cue, provides valid information on where a target will occur.

II. NETWORK CONNECTIVITY

Neural areas found active in studies of functional anatomy must be orchestrated in carrying out any real task. One approach to studying this connectivity uses fMRI to study the time course of activity and the correlations between active areas. Because of the relatively long delays between input and peak BOLD fMRI signal, small time differences may be hard to detect. Another approach to the examination of temporal connections between brain areas is based on electrical or magnetic signals since these signals can give higher temporal resolution, and can be combined with MRI to improve the spatial localization. A third approach to the measurement of connectivity involves the measurement of fiber tracts that connect neural areas by use of diffusion tensor imaging (DTI) that traces white matter tracts. Below we illustrate these methods by primarily considering the connectivity of the dorsal anterior cingulate during performance of the Attention Network Test. The organization of anatomical areas in alerting and orienting is not fully known, but some promising beginnings have taken place. In alerting the source of the attention appears in the locus coeruleus (lc). Cells in the lc have two modes of processing. One mode is sustained and is perhaps related,

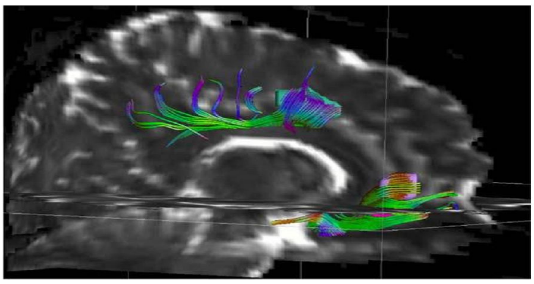

Fig. 2. The results of the Diffusion Tensor Imaging study of the structural connectivity of the dorsal and ventral anterior cingulate.

The colors indicate the orientation of the fibers red = left–right, green = anterior–posterior, and blue = inferior–superior. DTI images were acquired from a single subject on a 3.0 T Siemens Allegra MRI scanner. Diffusion weighting was per-3 formed using b = 700s mm along 60 independent orientations (Jone set al., 1999). MR acquisition parameters were: TE/TR = 110 ms/10.9 s; matrix = 128 × 128 on a 256 mm FOV, slice thickness = 2 mm with no gap; 60 transverse slices covering entire brain. DTI data were analyzed on a Siemens Leonardo TM workstation NUMARIS 4 satellite console using MGH's DTI Task Card 1.69 software written by R.P.

Wang (http://www.nmr.mhg.harvard.edu/˜rpwang/siemens/dti taskcard/new). (For in terpretation of the references to colour in this figure legend, the reader is referred to the web version of this article.) to the tonic level of alertness over long time intervals.

This function is known to involve the right cerebral hemisphere more strongly than the left (Coull et al. , 2000 ; Sturm & Willmess, 2001). Alertness is in uenced by sensory events and by the diurnal rhythm. However, its voluntary maintenance during task performance may be orchestrated from the anterior cingulated (Mottaghy et al., 2006). More phasic shifts of alerting can result from presenting any environmental signal. However, if the signal is likely to warn about an impending target this shift results in a characteristic suppression of the intrinsic brain rhythms (e.g. alpha) within a few tens of milliseconds and a strong negative wave (contingent negative variation) recorded from surface electrodes and that moves from a frontal generator toward the sensory areas of the hemisphere opposite the expected target. According to Bush, Luu, and Posner (2000) an analysis of a number of con ict tasks shows that the more dorsal part of the anterior cingulate is involved in the regulation of cognitive tasks, while the more ventral part of the cingulate is involved in regulation of emotion. One way to examine this issue is to image the structural connections of different parts of the cingulate using diffusion tensor imaging.

This form of imaging uses the diffusion of water molecules in particular directions due to the presence of militated fibers (Conturo et al., 1999). Thus it provides a way of examining the physical connections present in the brains of people. Fig. 2 shows the result of a DTI analysis of one subject run in our experiments. Note that the dorsal part of the ACC shows connections to cortical areas of the parietal and frontal lobes, while the ventral part of the ACC has strong connections to subcortical limbic areas (Abdullaev, in preparation).

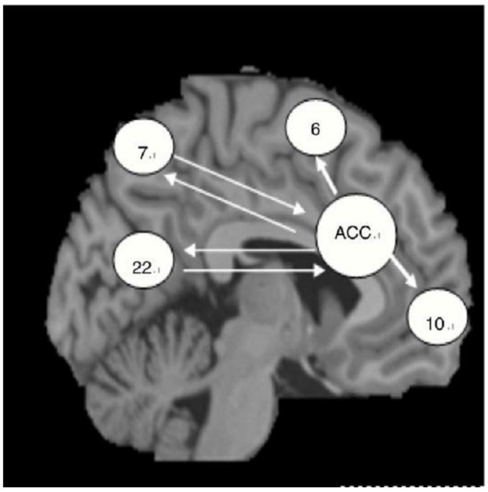

Fig. 3. The results of an fMRI connectivity analysis based on the correlations between the dorsal anterior cingulate and other cortical brain areas. Dynamic causal modeling (DCM) was used to infer the irection of in uence. Each circle contains the rodmann area involved. All in uences among regions shown by lines are significant in the direction of the arrow.

Another step is to examine networks of neural areas related to the ACC during task performance. To illustrate this approach we consider a recent fMRI study, in which we ran 12 young adults in the ANT. The data were first analyzed by statistical parameter mapping (SPM2) (http://www.fil.ion.ucl.ac.uk/spm) to produce the functional anatomy of different attentional circuits shown in Fig. 1 . For illustrative purposes we examined the cortical connectivity (Horwitz, Rumsey, & Donohue, 1998 ; Tang et al. , 2006) to and from the dorsal anterior cingulate during performance of the con ict subtraction in the ANT (incongruent–congruent). As expected we found strong connectivity to parietal and frontal brain areas. We then applied the Dynamic Casual Modeling (DCM) method for region of interest (ROI)-based effective connectivity analyses in the con ict task (Friston, Harrison, & Penny, 2003; Penny, Stephan, Mechelli, & Friston, 2004). Fig. 3 shows the results (numbers indicate Brodmann areas). Our findings showed interregional coupling (effective connectivity) for the con ict subtraction among the nodes of the attention system including the dorsal ACC, the lateral ventral prefrontal lobe (BA6, BA10), superior parietal gyrus (BA7) and temporal parietal junction (BA22). The ACC is likely to be the core mediation for the other brain regions in the con ict task. At the same time, BA7 and BA22 also modulate ACC. The interaction of the orienting and con ict network arises because following some cues (e.g. central) a shift of orienting must take place after the target occurs. Another way to examine network activity during the ANT is to use scalp EEG electrodes to record neural activity synchronized in different frequency bands. This method can be used to separate rapid temporal events, for example, it can separate the cue effects from the target effects in the ANT. In one study using the ANT (Fan et al., in preparation), a spatial cue indicating the location of the target produced increased high frequency gamma activity (above 30 Hz) about 200 ms after the cue presentation. When the cue brought attention to the target location, gamma activity was found following the cue, but not following the target. When the cue indicated the center location so that a shift of attention was needed following the target, the gamma activity was present following the target. These data suggest that gamma activity is associated with orienting of attention. It may occur 200 ms after the cue or only after the target depending upon when attention shifts. Taken together the fMRI, EEG and DTI methods can provide a detailed account of the orchestration of neural networks related to attention.

III. DEVELOPMENT OF ATTENTIONAL NETWORKS

The goal of understanding networks is to illuminate their role in actual human behavior. Efforts have been made to examine how the development of attentional networks in uences infant and child behavior. Such landmarks of development are crucial for links between genetic differences and actual behavior. The alerting and orienting systems begin development in early infancy and allow the infant to stay alert and to be in contact with sensory information. However, in this paper we concentrate on the executive network, which has been more difficult to demonstrate even in a rudimentary form during infancy. One effort involves anticipatory looking paradigms such as the Visual Expectation Paradigm (Haith, Hazan, & Goodman, 1988). In this method a repeating predictable sequence of visual stimuli is shown to znfants.Infant eye movements are recorded and coded for evidence of reactive looks, which occur in response to the presentation of stimuli, and anticipatory looks, which occur in advance of the presentation of stimuli. Reactive looks are thought to react exogenous control of attention in that looks to stimuli are in response to the stimuli itself and only require intentional processes associated with alerting and orienting. In contrast, anticipatory looks involve internal control of attention and may re ect the early functioning of the executive attention network. Anticipatory looking to more complex, ambiguous sequences of visual stimuli may present a method for examining more sophisticated forms of executive attention in infancy. Working with adults, Curran and Keele (1993) showed thatwhile stimuli appearing in an unambiguous sequence of locations (e.g., 123123) could be learned in the absence of attention, learning context-dependent sequences (e.g., 12131213)was dependent upon higher-order attentional processes. Clohessy, Posner, and Rothbart (2001) proposed that higher-order attentional processes may be required when anticipating a stimulus following Location 1 in a context-dependent sequence because there is a con ict that must be resolved between shifting attention to Location 2 and Location 3. They examined anticipatory looking to both unambiguous and context-dependent sequences of visual stimuli in infancy and found anticipatory looking during unambiguous sequences as early as 4 months. However, anticipatory looking during context-dependent sequences was not seen consistently until at least 18 months of age. These data suggest the possibility that rudimentary executive attention capacities may emerge during the first year of life but that more advanced con ict resolution capacities are not present until 2 years of age.

We are currently examining how executive attention as assessed through anticipatory looking is related to emotional and behavioral regulation in 7–9 month old infants. Preliminary analyses suggest that infants showing higher levels of anticipatory looking are also more likely to regulate approach tendencies when presented with novel toys and are also more likely to show self-soothing behaviors when presented with frightening stimuli. These results are consistent with findings from childhood that show higher levels of executive attention are broadly related to behavioral and emotional regulation (see Rothbart and Sheese (in press) , for a review). Another approach to examining executive attention in infancy involves the ability to detect errors (Berger, Tzur, &Posner, 2006) examined error detection capacities in seven month-old infants. Using a method developed by Wynn (1992), infants were presented with simple addition problems using a visual display of cartoon-like characters that were either correct (1 + 1 = 2) or incorrect (1 + 1 = 1). As in previous studies the infants looked longer at problems with incorrect answers. EEG analysis showed an increased frontal negativity for the incorrect problems that closely resembled that found in adults and is known to arise in the anterior cingulate. This finding suggests that executive attention in infancy arises in the same anatomy as found in adults. Two-year-olds have sufficient verbal and motor skills that have developed to allow for laboratory tests of executive attention that more closely resemble adult assessments.

A spatial con ict task using a touch-sensitive screen for responding has been used to examine con ict resolution with children as young as 2 years of age. The spatial con ict task developed by Gerardi-Caulton (2000) places an object’s identity and spatial location in con ict. At 24 months, children are generally unable to resolve this con ict and show high levels of incorrect responding. By 30 months children show much higher levels of correct responding but still show delays in reaction time on incongruent trials similar to what is found in adults. Children with better con ict resolution in the spatial-con ict task had higher levels of anticipatory looking and also higher parent-ratings of effortful control (Rothbart, Ellis, Rueda, & Posner, 2003). These results show the methodologically distinct measures of con ict resolution do re ect a common process that can be observed by parents in daily life.

The Child Attention Network Test (ANT-C) is a modified version of the adult ANT for use with children as young as 4 years of age. Using the ANT-C we (Rueda, Fan, et al., 2004) examined executive attention in children between 6 and 10 years of age and compared their performance with adults.

Results suggest that executive attention continues to develop throughout childhood but may stabilize at near-adult levels of performance by about eight years of age. There is considerable evidence that the executive attention network is of great importance in the acquisition of school subjects such as literacy (McCandliss, Beck, Sandak, & Perfetti, 2003), numeracy and in a wide variety of other subjects(Posner & Rothbart, 2007).

The impact of individual differences in executive attention can be seen in different areas of social development. We have proposed that individual differences in effortful control re ect differences in the functioning of the executive attention network (Posner & Rothbart, 1998 ; Rothbart, Derryberry, & Posner, 1994). Consistent with this hypothesis, we have found evidence that effortful control is related to executive attention measures throughout childhood (Gerardi-Caulton, 2000 ; Rothbart et al. , 2003). Effortful control, in turn, has been related to a broad range of outcomes relevant to social development including empathy, the regulation of negative affect, conscience development, and theory of mind (Rothbart& Bates, 2006). Emotion, thought, and behavior form a cluster of temporally associated processes in specific situations as experienced by the child. Single and repeated life experiences will thus shape connections between elicited emotion, conceptual understanding of events, and use of coping strategies to deal with these events. Several theorists have made contributions to this approach (e.g., Epstein (1998) and Mischel and Ayduk (2004)), but the overall framework is in keeping with the idea of Hebbian learning through network activation. Mischel and his colleagues have recently developed a cognitive affective personality (CAP) theory, making use of Cognitive Affective Units (CAUs) seen to operate within a connectionist network (Mischel & Ayduk, 2004). In their model, CAUs are variables encoding the features of situations, which include environmental effects as well as self-initiated thoughts.

IV. GENES AND EXPERIENCE BUILD NETWORKS

As more is known about the developmental progression of executive attention as discussed above, there is an increased possibility of accounting for both the general development of the network and individual differences by examining how genes and experience interact to shape the executive attention network. Some progress made in that direction is discussed below.

To determine genes that might be related to building an attentional network we used the Attention Network Test (ANT) to examine individual differences in the efficiency ofexecutive attention. We first used the ANT to assess attention in monozygotic and dizygotic same-sex twins (Fan, Wu, Fossella, & Posner, 2001). We found strong heritability of the executive network. These data supported a search for genes in executive attention.

We then used the association of the executive network with the neuro modulator dopamine as a way of searching for candidate genes that might relate to the efficiency of the network (Fossella et al., 2002). To do this, 200 persons performed the ANT and were genotyped to examine frequent polymorphisms in genes related to dopamine. We found significant association of two genes, the dopamine D4 receptor (DRD4) gene and monoamine oxidase a (MAOA) gene, with executive attention. We then conducted a neuroimaging experiment in which persons with different alleles of these two genes were compared while they performed the ANT (Fan, Fossella, Sommer, & Posner, 2003). Groups with different alleles of these genes showed differences in the ability to resolve con ict as measured by the ANT and also produced significantly different activations in the anterior cingulate, a major node of the executive attention network. Recent studies have extended these observations. In two different studies employing con ict related tasks other than the ANT, alleles of the catechol-o-methyl transferase (COMT) gene were related to the ability to resolve con ict (Blasi et al.2005 ; Diamond, Briand, Fossella, & Gehlbach, 2004). A study using the child ANT showed a significant relation between the dopamine transporter (DAT1) and executive attention as measured by the ANT (Rueda, Rothbart, McCandliss, Saccamanno, & Posner, 2005). In addition, research has suggested that genes related to serotonin transmission alsoin uence executive attention (Canli et al., 2005 ; Reuter, Ott, Vaidl, & Henning, in press).

The relation of genetic factors to the functioning of the executive attention system does not mean that the system cannot be in uenced by experience. Several training-oriented programs have been successful in improving attention in patients suffering from different pathologies. For example, the use of Attention Process Training (APT) has led to specific improvements in executive attention in patients with specific brain injury (Sohlberg, McLaughlin, Pavese, Heidrich,& Posner, 2000) as well as in children with

Attention Deficit Hyperactivity Disorder (ADHD) (Kerns, Esso, & Thompson, 1999). Work with ADHD children has also shown that working memory training can improve attention (Klingberg, Forssberg, & Westerberg, 2002; Olesen, Westerberg, & Klingberg, 2004 With normal adults, training with video-games produced better performance on a range of visual attention tasks (Green & Bavelier, 2003).

To examine the role of experience on the executive attention network we have developed and tested a five-day training intervention that uses computerized exercises. We tested the effect of training during the period of major development of executive attention, which takes place between 4 and 7 years of age according to our previous results (Rueda, Fan, et al., 2004 We hoped to observe an improvement in con ict resolution as measured by the ANT in trained children, along with changes in the underlying network and generalization to other aspects of cognition. EEG data showed clear evidence of improvement in network efficiency in resolving con ict following training (Rueda et al., 2005). The N2 component of the scalp recorded ERP has been shown to arise in the anterior cingulate and is related to the resolution of con ict (Rueda , Posner , et al. , 2004 van Veen & Carter, 2002). We found N2 differences between congruent and incongruent trials of the ANT in trained six-year-olds, that resembled differences found in adults. In the four year olds training seemed to in uence more anterior electrodes related to emotional control areas of the cingulate (Bush et al., 2000). These data suggest that training altered the network for the resolution of con ict in the direction of being more like what is found in adults.

We also found a significantly greater improvement in intelligence in the trained group compared to the control children. This finding suggested that training effects had generalized to a measure of cognitive processing that is far removed from the training exercises. We did not observe changes in temperament over the course of the training, but this was expected, due to the short time elapsing between assessment sessions.

Not all children need or benefit from attention training. This may be why variability is so high in children's performance. In some of our studies, children with the most initial difficulty in resolving con ict showed the greatest overall improvement due to training. Our research has also suggested a genetic marker of initial differences in attention among the children. We were able to genotype most of the 6-year-old children participating in our training study. Children were divided into two groups according to their particular form of a genetic polymorphism in the 3 untranslated region of the dopamine transporter (DAT1); those carrying the pure long form and those carrying the mixed short/long form of the gene. Since our sample was small, we combined 6-year-olds who had attention training with those in the control condition. Although there were only seven children in the pure long allele group and eight in the mixed long/short group, we found a significantly greater efficiency in con ict scores for the pure long allele group.

Several features of our data supported the relation between the DAT1 polymorphism and individual differences in the efficiency of executive attention. Children in the two groups differed in their con ict scores on the ANT as well as in the effortful control scores obtained from the Children's Behavior Questionnaire (Rothbart, Ahadi, Hershey, & Fisher, 2001). In particular, the short/long mixed group showed higher con ict scores and lower effortful control than those in the pure long group. The two groups also differed in their EEG data. In the first session, children with the pure long allele showed the effect of ankers in the expected direction (larger N2 for incongruent trials), whereas children in the mixed alleles group did not show this effect. The larger N2 for incongruent trials was also found for trained children of 6 years of age and adults. Thus the presence of the pure long allele is associated with more mature executive attention. The DAT1 gene has also been associated with attention deficit hyperactivity disorder (ADHD). However, the exact relation between executive attention efficiency in normals and the presence of attention deficits in ADHD is not clear (Swanson et al., 2000).

Our findings are preliminary, because of the small number of children involved. We are currently examining a larger cohort of children in a longitudinal study from 7 months to 4 years of age.

We hope to replicate our current results and explore other genes that might in uence the development of attentional networks. We also hope to explore the origin in infancy of the executive attention network that we have measured in childhood. Given the wide range of individual differences in the efficiency of attention, it is expected that attention training could be especially beneficial for those children with poorer initial efficiency. These could be children with pathologies that involve attentional networks, children with genetic backgrounds associated with poorer attentional performance, or children raised in different degrees of deprivation.

Genes do not directly produce attention. What they do is code for different proteins that in uence the efficiency with which modulators such as dopamine are produced and/or bind to their receptors. These modulators are in turn related to individual difference in the efficiency of the attention networks. There is a great deal in common among humans in the anatomy of high level networks, and this must have a basis within the human genome. The same genes that are related to individual differences in attention are also likely to be important in the development of the attentional networks that are common to all humans. Some of these networks are also common to non - human animals. By examining these networks in animals it should be possible to understand the role of genes in shaping networks.

Can animals perform the same tasks we have developed for humans? The answer is clearly yes. Monkeys have been trained to shift attention to cues and to carry out con ict tasks like those in the ANT. More recently rodents have also been trained in attention shifting tasks (Beane & Marrocco, 2004).These tasks make it possible to examine the role that genes play in carrying out the same attentional operations as have been studied in humans.

It has also been reported that areas of the frontal midline corresponding to the anterior cingulate are activated in the mouse during trace but not delayed conditioning (Han, O'Tuathaigh, & Koch, 2004). Since trace and delayed conditioning are both very simple tasks and the two are quite similar they could be used to assay operation of rodent brain areas that may be related to executive attention in humans. An important need in this effort is the development of methods to manipulate relevant genes in specific anatomical locations that are important nodes of a particular network.

Usually genes are expressed at multiple locations, so that changes (e.g. knock out studies) are not specific to one brain area. However, using subtractive genomics, a method currently being developed (Dumas et al., 2005), it should become possible to determine the specific operations performed by genes at particular places in different attentional networks.

We believe that this kind of genetic analysis of network development will create a productive link between genes and both normal and pathological psychological function.

REFERENCES

[1]. Abdullaev, Y. Connectivity of the anterior cingulate. Unpublished study, University of Oregon Lewis Center for Neuroimaging (in preparation).

[2]. Beane, M., & Marrocco, R. (2004). Cholinergic and noradrenergic inputs to the posterior parietal cortex modulate the components of exogenous attention.

[3]. In M. I. Posner (Ed.), Cognitive neuroscience of attention (pp. 313–325). New York: Guilford.

[4]. Berger, A., Tzur, G., & Posner, M. I. (2006). Infant brains detect arithmetic errors. Proceedings of the National Academy of Sciences of the United States of America, 103, 12649–12653.

[5]. Blasi, G., Mattay, G. S., Bertolino, A., Elvev° ag, B., Callicott, J. H., Das, S., 1 58 et al. (2005). Effect of cCatechol- O -Methyltransferase val met genotype on attentional control. Journal of Neuroscience, 25, 5038–5045.

[6]. Bullock, T. H., Bennett, M. V. L., Johnston, D., Josephson, R., Marder, E., & Fields, R. D. (2005). The neuron doctrine, redux. Science, 310, 791–793.

[7]. Bush, G., Luu, P., & Posner, M. I. (2000). Cognitive and emotional in uences in the anterior cingulate cortex. Trends in Cognitive Science, 4–6, 215–222.

[8]. Canli, T., Omura, K., Haas, B. W., Fallgatter, A., Todd, R., Constable, R. T., et al. (2005). Beyond affect: A role for genetic variation of the serotonin transporter in neural activation during a cognitive attention task. Proceedings of the National Academy of Sciences of the United States of America, 102, 12224–12229.

[9]. Clohessy, A. B., Posner, M. I., & Rothbart, M. K. (2001). Development of the functional visual field. Acta Psychologica, 106 , 51–68.

[10]. Conturo, T. E., Lori, N. F., Cull, T. S., Akbudak, E., Snyder, A. Z., Shimony, J. S., et al. (1999). Tracking neuronal fiber pathways in the living human brain. Proceedings of the NationalAcademy of Sciences of the United States of America, 96(18), 10422–10427.

[11]. Corbetta, M., & Shulman, G. L. (2002). Control of goal-directed and stimulus driven attention in the brain. Nature Neuroscience Reviews, 3, 201–215.

[12]. Coull, J. T., Frith, C. D., Buchel, C., &Nobre, A. C. (2000). Orienting attention in time: Behavioural and neuroanatomical distinction between exogenous and endogenous shifts. Neuropsychologia, 38(6), 808–819.

[13]. Crottaz-Herbtte, S., & Mennon, V. (2006). Where and when the anterior cingulate cortex modulates attentional response: Combined fMRI and ERP evidence. Journal of Cognitive Neuroscience, 18, 766–780.

[14]. Curran, T., & Keele, S. W. (1993). Attentional and non-attentional forms of sequence learning. Journal of Experimental Psychology: Learning, Memory, and Cognition, 19, 189–202.

[15]. Diamond, A., Briand, L., Fossella, J., & Gehlbach, L. (2004). Genetic and neurochemical modulation of prefrontal cognitive functions in children. American Journal of Psychiatry, 161, 125–132.

[16]. Dumas, T., Hostick, U., Wu, H., Spaltenstein, J., Ghatak, C., Nguyen, J., et al.(2005). Maximizing the anatomical specificity of native neuronal promoters by a subtractive transgenic technique. Society for Neuroscience Abstracts.

[17]. Epstein, S. (1998). Cognitive-experiential self-theory: A dual process personality theory with implications for diagnosis and psychotherapy.

[18]. In R. F. Bornstein, & J. M. Masling (Eds.), Empirical perspectives on the psychoanalytic unconscious: vol. 7 (pp. 99–140). Washington, DC: American Psychological Association.

[19]. Etkin, A., Egner, T., Peraza, D. M., Kandel, E. R., & Hirsch, J. (2006). Resolving emotional con ict: A role for the rostralanterior cingulate cortex in modulating activity in the amygdala. Neuron, 51, 871–882.

[20]. Fan, F., Byrne, J., Worden, M. S., McCandliss, B. D., Fossella, J., & Posner, M.I. The relation of brain oscillations to attentional networks (in preparation).

[21]. Fan, J., Fossella, J. A., Sommer, T., & Posner, M. I. (2003). Mapping the genetic variation of executive attention onto brain activity. Proceedings of the National Academy of Sciences of the United States of America, 100, 7406–7411.

[22]. Fan, J., McCandliss, B. D., Fossella, J., Flombaum, J. I., & Posner, M. I. (2005). The activation of attentional networks. Neuroimage, 26, 471–479.

[23]. Fan, J., McCandliss, B. D., Sommer, T., Raz, M., & Posner, M. I. (2002).Testing the efficiency and independence of attentional networks. Journal of Cognitive Neuroscience, 3(14), 340–347.

[24]. Fan, J., Wu, Y., Fossella, J., & Posner, M. I. (2001). Assessing the heritability of attentional networks. Biomedical Central Neuroscience, 2, 14.

[25]. Fossella, J., Sommer, T., Fan, J., Wu, Y., Swanson, J. M., Pfaff, D. W., et al. (2002). Assessing the molecular genetics of attention networks. BMC Neuroscience, 3, 14.

[26]. Friston, K. J., Harrison, L., & Penny, W. (2003). Dynamic causal modeling. Neuroimage, 19(4), 1273–1302.

[27]. Gerardi-Caulton, G. (2000). Sensitivity to spatial con ict and the development of self-regulation in children 24–36 months of age. Developmental Science, 3–4, 397–404.

[28]. Green, C. S., & Bavelier, D. (2003). Action video game modifies visual attention. Nature, 423, 534–537.

[29]. Haith, M. M., Hazan, C., & Goodman, G. S. (1988). Expectation and anticipation of dynamic visual events by 3.5 month old babies. Child Development, 59, 467–469.

[30]. Han, C. J., O'Tuathaigh, C. M., & Koch, C. (2004). A practical assay for attention in mice. In M. I. Posner (Ed.), Cognitive neuroscience of attention (pp. 294–312). New York: Guilford.

[31]. Hillyard, S. A., Di Russo, F., & Martinez, A. (2004). The imaging of visual attention. In N. Kanwisher, & J. Duncan (Eds.), Functdional neuroimaging of visual cognition attention & performance XX (pp. 381–390). Oxford, UK: Oxford University Press.

TRAFFIC ANALYSIS AND ENERGY OPTIMIZATION THROUGH DUTY CYCLE MANAGEMENT IN WIRELESS SENSOR NETWORK

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Abstract- A WSN can be deployed in remote geographical locations and requires minimal setup and overall administration costs. Sensor nodes are hardware devices and their source of energy is battery power . Nodes store , forward , report various environment related parameters to the sink , which is normally a base station ,thus monitor the environment using co-operative information. Duty cycling is the ratio of active time i.e time at which the particular set of nodes are active to the whole scheduling time. In this scheme, each node alternates between active and sleeping states so the nodes need not to be active or in waiting period all the time during its complete lifespan. Thus, saves energy. In this paper, energy consumption is reduced in case of broadcasting. Also, link quality is also maintained which reduces the path failure due to poor link quality. In addition to the above, it analyses, monitors and handles traffic caused due to heavy congestion. Thus congestion, delay can be controlled which improves efficiency and overall performance of the network.

Keywords: *ADB, Duty Cycle, Link Quality, Multihop, Scheduling, Sensor Node, Sink Node, Wireless Sensor Network.*

I. INTRODUCTION

A. Wireless Sensor Network

A wireless sensor network consists of group of sensors, or nodes that are linked by a wireless medium to perform distributed sensing tasks. The sensors are assumed to have a fixed communication and a fixed sensing range, which can significantly vary depending on the type of sensing, performed. It has received a greater interest in various applications such as disaster management, border protection, combat field reconnaissance, in military for security surveillance, structural health monitoring, industrial automation, civil structure monitoring, and monitoring the biologically hazardous places and in variety of applications.

A sensor network must be able to operate under very dynamic conditions. Specifically, our protocols must be able to enable network operation during start-up, steady state, and failure. The necessity of operation under these conditions is required because in most cases, the sensor network must operate unattended. Once the nodes have booted up and a network is formed, most of the nodes will be able to sustain a steady state of operation, i.e. their energy reservoirs are nearly full and they can support all the sensing, signal processing and communications tasks as required. In this mode, the bulk of the nodes will be formed into a multi-hop network. The node begin to establish routes by which information is passed to one or more sink nodes.

B. Sink node

A Sink node is similar to head node which gather, control data collected by other sensor node. Also it is a sensor node with gateway functions to link to external networks such as the Internet and sensed - information is normally distributed via the sink node.A sink node may be a long-range radio, capable of connecting the sensor network to existing long haul communications infrastructure. The sink may also be a mobile node acting as an information sink, or any other entity that is required to extract information from the sensor network. Although the multi-hop network can operate in both the sensor-to-sink or sink-to-sensor i.e broadcast or multi-cast modes. This will put significant strain on the energy resources of the nodes near the sink, making that neighborhood more susceptible to energy depletion and failure.

Nodes may fail due to other reasons such as mechanical failure. When many nodes have failed, the MAC and routing protocols must accommodate formation of new links and routes to the sink nodes. This may require actively adjusting transmit powers and signaling rates on the existing links to reduce energy consumption, or rerouting packets through regions of the network where nodes have more energy left.

C. Sensor Node

Sensor nodes are expected to operate autonomously in unattended environments and potentially in large numbers. Failures are susceptible in wireless sensor networks due to inhospitable, unstable environment and unattended deployment. The data communication and various network operations cause energy depletion in sensor nodes and therefore, it is common for sensor nodes to exhaust its energy completely and stop operating. This may cause connectivity and data loss during communication. Therefore, it is necessary that network failures are detected in advance and appropriate measures should be taken to sustain network operation. Connections to establish communication between nodes may be formed using media such as infrared devices or radios. Figure 1. shows the components of sensor node

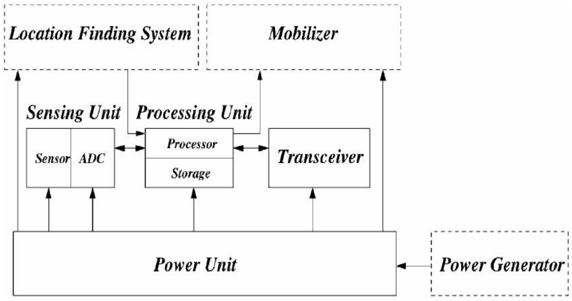


Figure 1. Mechanism of Sensor Node

A single Wireless Integrated Network Sensor (WINS) node combines micro-sensor technology, low power signal processing, low power computation, low power, and low cost wireless networking capability in a compact system.

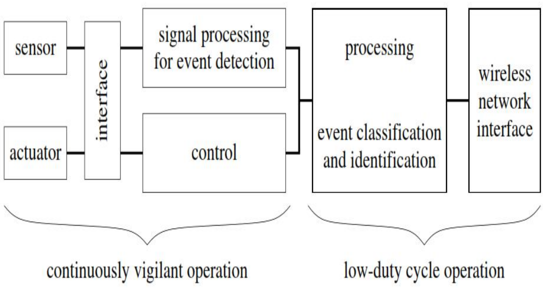


Figure 2. Node Architecture

Figure 2. gives a description of the WINS node architecture[2].The stationary node will maintain a registry as well evenif its role is minimal compared to that of the mobile node. The stationary node will register mobile sensors that have formed connections and remove them when the link is broken thus, effectively limiting participation in the connection procedures. To design a system in which the mobile assumes full responsibility of making and breaking connections, a ideal signaling method must be defined. Figure 3. shows the architecture of sensor network.

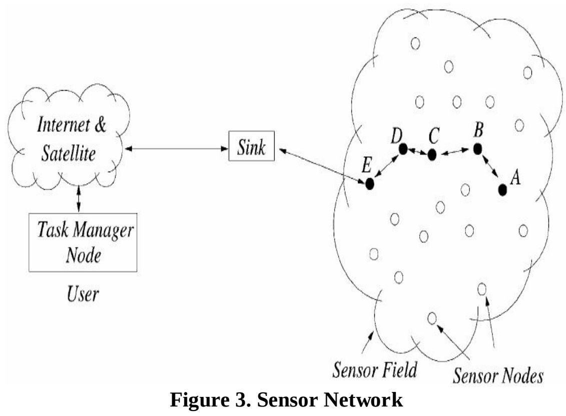


Figure 3. Sensor Network

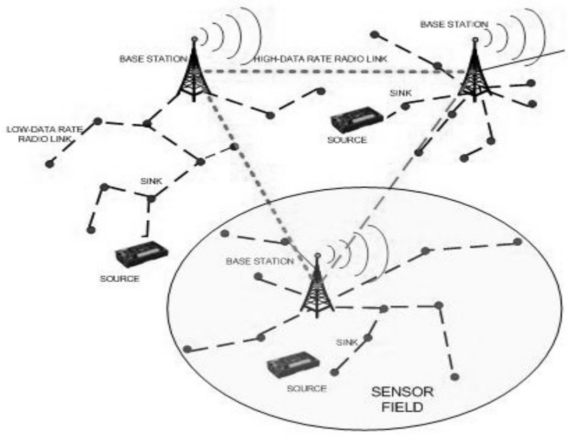


Figure 4. Architecture of Wireless Sensor Network

II. DUTY CYCLE MANAGEMENT USING ADB PROTOCOL

A. *Duty Cycle Approaches*

Wireless sensor devices can use duty cycling techniques to increases battery lifespan by adapting their active duration based on network traffic. Nodes can also regulate data rate and transmission power to reduce energy consumption while maintaining the required signal quality. Duty Cycle (DC) approaches can be grouped into: asynchronous DC ; synchronised or scheduled DC; and hybrid approaches. Asynchronous duty cycling (ADC) is typified by Low Power Listening (LPL) and B-MAC. The radio is turned on for very short amounts of time to check for channel activity known as channel polling. If activity is detected the radio remains on to receive data, else it turns off. Transmitting nodes precede messages with a preamble longer than the sleep time of the recipient, to guarantee they will have turned their radios on, detected the channel activity and be ready to receive before the preamble is ended. This places an energy cost on the transmitter, more so for DC rates using longer off periods. Long preambles can also increase network congestion.

X-MAC attempts to lessen the transmit burden by having receivers send an acknowledgement (ACK) as soon as they detect channel activity to cut the preamble short and start transmitting the data. Such optimisations are not beneficial to broadcasts as neighbouring nodes may have a wide range of different on times and a full-length preamble will be necessary. Leading examples of synchronised DC schemes include SMAC and T-MAC, which maintain and synchronise schedules with neighbouring nodes to record when each node is going to be turned on.

Periodic control messages advertise a node's schedule to neighbours during contention periods, which are considerably longer than those for channel polling. This becomes significant during periods of inactivity when energy is spent listening when there is nothing to receive. TMAC uses a threshold time in this period after which, if no transmissions have been detected the node turns off early. ZMAC also uses scheduling, but allows nodes to "steal" each-others slots under certain conditions. Hybrid schemes combine synchronisation, schedules and preambles for use during communication.

MAC nodes maintain schedules of when neighbours are likely to be on, allowing shorter preambles to be used that do not cover the entire off period. In a different approach, MH-MAC switches between asynchronous and synchronous modes of operation, thereby enabling contention during periods of lighter congestion and contention-free communications during periods of heavy network usage. SCP-MAC uses synchronised channel polling allowing the use of short preambles. The various duty cycle levels are shown in Figure 5.

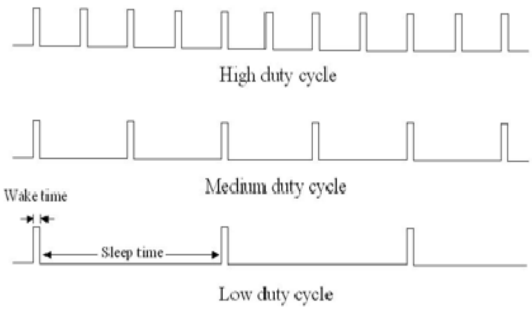


Figure 5. Multiple duty cycle levels

B. ADB Protocol

ADB is the Asynchronous Duty-cycle Broadcasting protocol for efficient multihop broadcast in wireless sensor networks using asynchronous duty-cycling. ADB dynamically optimizes the broadcast at the level of transmission to each individual neighbor of a node, as the neighbors asynchronously wakeup. ADB is integrated with the MAC layer to exploit information only available at this layer. ADB takes advantage of the fact that nodes wake up at different times to optimize the progress of a multihop broadcast at a finer granularity.

Instead of treating transmission from a node to all of its neighbors as a basic unit of progress for the broadcast, ADB optimizes at the level of transmission to each neighbor individually. As neighbors wake up at different times, a sender with ADB uses unicast to reach each neighbor, so that the sender accurately learns which neighbors have been reached by the broadcast and improving reliability through the use of Automatic Repeat Request (ARQ) as part of the unicast transmission.

The sender also updates each receiver with information on the progress of the broadcast, helping a node avoid redundant transmissions and allowing delegating transmission for some neighbor to another neighbor with better link quality to it. This approach allows a node to sleep as early as possible and avoids transmissions over poor links, further reducing energy consumption and delivery latency.

❖ *Features of ADB Protocol*

- ADB allows a node to go to sleep once all its neighbors have been reached or been delegated to other nodes
- ADB delivers a broadcast packet without occupying the medium while waiting for each receiver to wake up, to allow a neighbor to start rebroadcasting the packet immediately
- ADB attempts to avoid transmissions over poor links
- ADB informs a neighbor that has just waken up on the progress of a broadcast, to avoid unnecessary waiting and transmissions.

❖ *Working of ADB Protocol*

Figure 3 gives an overview of the operation of ADB. In this simple example, the network consists of three nodes, nodes *S*, *R1*, and *R2*, all within transmission ranges of each other. Node *S* wants to broadcast a DATA packet to all nodes. When *R1* wakes up, node *S* transmits the packet upon receiving *R1*'s beacon in the same way as for unicast in RI-MAC. However, ADB includes a new "footer" in DATA frames and acknowledgment beacons (ACKs), indicating the progress of the broadcast, including some transmissions that are about to happen. A receiving node uses this information to avoid unnecessary transmissions and to decide whether it should forward the packet to a neighbor that has not received it.

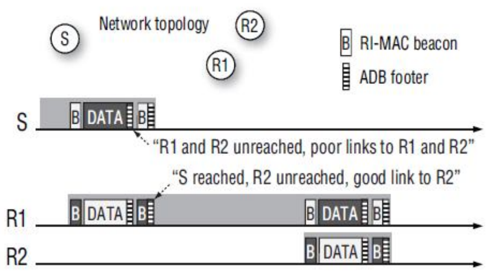


Figure 6. Overview of ADB Protocol

In this example, the ADB footer in the DATA frame from *S* informs *R1* that *R2* has not been reached yet by the broadcast and that the quality of the link (*S*,*R*2) is poor. Suppose the quality of link (*R*1, *R*2) is good (e.g., because of the short distance). Node *R1* decides to delivery the packet to *R2* and indicates the good quality of (*R*1, *R*2) in the footer of the ACK to *R1*. Upon receiving this ACK, *S* learns that it is better for *R1* to transmit the packet to *R2*, so *S* "*delegates*" handling of *R2* to *R1*. As *S* has no other neighbor to be reached, *S* then goes to sleep immediately. When

R2 wakes up, *R1* unicasts the DATA frame to *R2* in the same way, except that the ADB footer in the DATA frame indicates that *S* has received the DATA frame, allowing *R2* to sleep immediately because all neighbors of *R2* have been reached.

III. IMPORTANCE OF LINK QUALITY IN WIRELESS TRANSMISSION IN WSN

Our major design goal is to reduce the number of transmissions in the network and increase routing throughput by utilizing long-range bursty links for packet forwarding. To achieve this goal, we need to provide relevant support at both link estimation and routing level. The Packet Reception Rate (PRR) and Packet Loss Rate (PLR) are the most common of such indicators. Both PRR and PLR are based on a given number of packet transmissions. One of the most desirable features for a Link Quality (LQ) indicator is its ability to assess the channel by high degree of reliability by using the minimum possible resources in terms of time and energy.

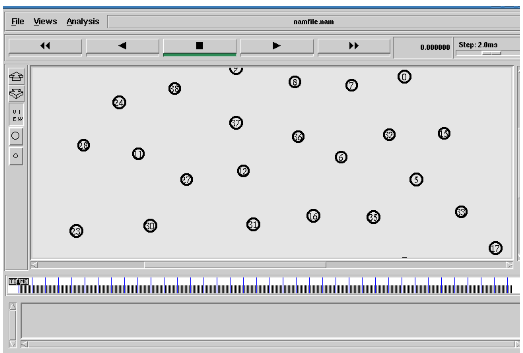
The link quality can be defined as:

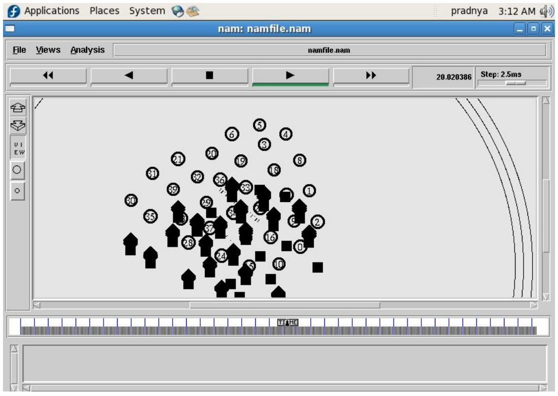
LQ Indicator = PRR£norm (meanRSSI):

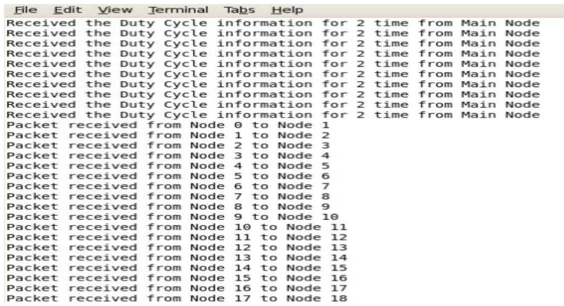
where, LQ->Link quality
PRR-> Packet Reception Rate
RSSI-> Received Signal Strength Indicator

Link quality can be summarized with the following points: (1)bursty links i.e the intermediate wireless link with a PRR between 10% and 90% as a *bursty link* if packet delivery on this link is correlated. This means that shifts between phases of reliable and poor packet delivery occur at short-time scales, but future packet delivery is correlated to the very recent success rate. Reliable links, that is, with a PRR >90%, can also be bursty can be used for packet forwarding during their stable periods without affecting the reliability and stability of existing routing protocols (2) these links often achieve significantly better routing progress and routing throughput than the long-term stable links chosen by existing routing protocols.

IV. SIMULATION RESULTS







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File  Edit  View  Terminal  Tabs  Help
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Received the Duty Cycle information for 2 time from Main Node
Packet received from Node 0 to Node 1
Packet received from Node 1 to Node 2
Packet received from Node 2 to Node 3
Packet received from Node 3 to Node 4
Packet received from Node 4 to Node 5
Packet received from Node 5 to Node 6
Packet received from Node 6 to Node 7
Packet received from Node 7 to Node 8
Packet received from Node 8 to Node 9
Packet received from Node 9 to Node 10
Packet received from Node 10 to Node 11
Packet received from Node 11 to Node 12
Packet received from Node 12 to Node 13
Packet received from Node 13 to Node 14
Packet received from Node 14 to Node 15
Packet received from Node 15 to Node 16
Packet received from Node 16 to Node 17
Packet received from Node 17 to Node 18
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V. CONCLUSION

In this paper, energy consumption is reduced in case of broadcasting, link quality is also maintained which reduces the path failure due to poor link quality, avoid congestion by monitoring, regulating traffic, reduces delays which increase throughput. Thus, improves the performance of the system through increasing the lifespan of the sensor nodes. Hence, enhance the feature of adaptivity.

REFERENCES

[1] Ms. J. V. Shiral, Asst. Prof. B. N. Mahajan, “Adaptive Duty Cycle Aware Broadcasting and Scheduling in Wireless Sensor Network”, International Journal of Engineering Research and Applications, Vol 2, pp. 2578-2584, May-Jun 2012.

[2] B. Murray, T. Baugé, R. Egan, C. Tan, C. Yong, “Dynamic Duty Cycle Control with Path and Zone Management in Wireless Sensor Networks,” IEEE International Conference on Wireless Communications and Mobile Computing, 2008.

[3] Katayoun Sohrabi, Jay Gao,“Protocols for Self-Organization for Wireless Sensor Network”, IEEE Journal on Personal Communication, 2000.

[4] Nandini. S. Patil, Prof. P. R. Patil, “Data Aggregation in Wireless Sensor Network,”IEEE International Conference on Computational Intelligence and Computing Research, 2010.

[5] Roberto Di Pietro, Pietro Michiardi, Refik Molva, “Confidentiality and Integrity for Data Aggregation in WSN using Peer Monitoring”, 2007.

[6] Abolfazl AkbariNeda Beikmahdavi and Bahman Shahamat Naderi, “Cluster-based and cellular approach to fault detection and recovery in wireless sensor network,”International Journal of Mobile and Wireless, Vol 2, No. , Feb 2010 .

[7] Shuang Li, Raghu Kisore Neelisetti, Cong Liu, and Alvin Lim, "Efficient Multipath Protocol for WSN", International Journal of Mobile and Wireless, Vol 2, No.1 , Feb 2010.

[8] Xianlong Jiao,Wei Lou,Junchao Ma," Minimum Latency Broadcast Scheduling in Duty-Cycled Multi-Hop Wireless Networks", IEEE Transaction on Parallel and Distributed Systems, June 2011.

[9] Jinbao Li; Desheng Zhang; Longjiang Guo,"DCM: A duty cycle based multi-channel MAC protocol for wireless sensor networks ", IET International Conference, 05 April 2011.

[10] Simarpreet Kaur, and Leena Mahajan,"Power Saving MAC Protocols for WSNs and Optimization of S-MAC Protocol", International Journal of Radio Frequency Identification and Wireless Sensor Networks, 02 June 2011.

[11] Fei Tong , Rong Xie , Lei Shu and Young-Chon Kim,"A Cross-Layer Duty Cycle MAC Protocol Supporting a Pipeline Feature for Wireless Sensor Networks", 2011, 11, 5183-5201.

[12] Long Cheng, *Canfeng Chen, Jian Ma, Lei Shu, Hongyang Chen and Laurence T. Yang*,"Residual Time Aware Forwarding for Randomly Duty-Cycled Wireless Sensor Networks", IEEE Transaction on Parallel and Distributed Computing , 05 April 2009.

[13] Hyung Seok Kim, *Member, Joo-Han Song, Member, and Seok Lee*,"Energy-Efficient Traffic Scheduling in IEEE 802.15.4 for Home Automation Networks"IEEE Transaction on Computers, June 2006.

[14] Heinrich Luecken, Thomas Zasowski, and Armin Wittneben," Synchronization Scheme for Low Duty Cycle UWB Impulse Radio Receiver"IEEE Transaction on Communication, April 2008.

[15] John D. Lea-Cox, Andrew G.Ristvey, David S. Ross "A Low-Cost Multihop in Wireless Sensor Network, Enabling Real- Time Management of Environmental Data for the Green House and Nursery Industry ", GreenSys:High Technology for GreenHouse System Management, Int. Soc. Hort. Sci. Spec. Conf, 3-6 Oct 2007. Acta Hort, 2008 (In Press).

[16] Yenumula B. Reddy , Rastko Selmic "Secure Packet Transfer in Wireless Sensor Networks – A Trust-based Approach" IARIA, The Tenth International Conference on Networks, 2011.

[17] Xue Wang , Jun-Jie M , Sheng Wang and Dao-Wei Bi , "Prediction-based Dynamic Energy Management in Wireless Sensor Networks", ISSN, 7, 251-266, 2007.

[18] Muhammad Hamad Alizai, Olaf Landsiedel, and Klaus Wehrle. "Exploiting the Burstiness of Intermediate-Quality Wireless Links", International Journal of Distributed Sensor Networks, 2012.

[19] Michele Rondinone, Junaid Ansari, Janne Riihijarvi and Petri Mahonen "Designing a Reliable and Stable Link Quality Metric for Wireless Sensor Networks", REALWSN, 2008.

[20] ADB: An Efficient Multihop Broadcast Protocol based on Asynchronous Duty-cycling in Wireless Sensor Networks "Yanjun Sun, Omer Gurewitz, Shu Du, Lei Tang, David B. Johnson", SenSys'09.

[21] Wireless Sensor Network Communication Architecture for Wide-Area Large Scale Soil Moisture Estimation and Wetlands Monitoring "Miguel Angel Erazo Villegas, Seok Yee Tang, Yi Qian", Network Communications Infrastructure Group Department of Electrical and Computer Engineering University of Puerto Rico at Mayagüez.

[22] Joint Duty Cycle and Link Adaptation for IEEE 802.15.4 beacon-enabled networks "Rodolfo de Paz Alberola and Dirk Pesch" , HotEmNets 2010.

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CLOUD ENVIRONMENT FOR PROJECT ALLOCATION SYSTEM

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Abstract- The Student Project Allocation problem (SPA) is a many-to-one matching problem and is an extension of the extensively studied one-to-one Stable Marriage Problem (SMP[1]). Although a complete and comprehensive solution to the many-to-one matching problem has yet to be found, there are many comparable examples that offer a solution based on allowing a level of compromise within the results they obtain. It is easy to develop a website to maintain the allocation system online. Whenever we use online allocation system, we can face problems like low bandwidth, processing speed low, low frequency when lots of user is online, etc. We proposed to carry out the annual allocation of students to individual project by a centralised mechanism, taking as input, the preferences of students on the available projects. Design and create an application for allocating the Project to the students using Cloud Environment.

Keywords-*Cloud computing; Service integration [3]; Innovation project characteristics.*

I. INTRODUCTION

Within the Department of Computer Science at the University of Sheffield (UoS CS Department), dissertation projects are supervised by an academic whose specialism is broadly in the same field as the student's area of study. Each project has to have been deemed to suitably fulfill the dissertation criteria set out by the University, before it is approved and allowed to be undertaken. There are usually two ways of achieving this: firstly, academics within a department will come up with a set number of appropriate project proposals which students are then able to choose from; secondly, students can propose their own ideas for a project to an academic. In practice, a combination of both methods are usually adopted. Although in theory, these approaches to project allocation seem relatively simplistic, the reality of implementing such a system is far from straightforward[3]. Looking at the approach whereby academics propose projects for students to choose from, conflicts regularly arise wherein two or more students will be competing for the same desired project; it must then be decided which candidate the project will be allocated to. Whilst there are ways of resolving such a dilemma, it is inevitable that there will be students who won't get to undertake their project of choice, leaving some students dissatisfied[2]. The second approach would seem to address this issue by allowing the student to propose a project they would want to pursue. However, in practise, there are several constraints that this method imposes upon both student and academic; for example, a student should propose a project to an academic whose area of expertise lies in the scope of the students' desired project. An academic is only allowed to supervise a limited number of students and if those supervisors who had elected to undertake a student's proposed projects but had already filled their quota of students, then that supervisor would be unable to take on that student's self proposed project. using this cloud environment Cloud computing uses a mix of old and new concepts to support virtualization, on-demand services, scalable flexibility, hardware and software scalability, automatic adaptation, pay-per-use, and service level agreements, among other features Since it offers many computational alternatives, cloud computing is showing up in a growing number of IT services in the form of cloud utilities Cloud computing clearly exposes applications to clouds, and supports the use of a variety of online products and services such as Salesforce.com

There are a number of steps4 that can be followed which can be tailored to a wide variety of constraints and situations, forming a guideline to help achieve a logical, methodical and suitable approach to effective resource allocation:

1. Identify/design alternatives
2. Identify and structure organization's goals and objectives
3. Prioritize the objectives and sub-objectives
4. Measure each alternative's contribution to each of the lowest level sub objectives
5. Find the best combination of alternatives, subject to environmental and organizational constraints

II. WHY CLOUD ENVIRONMENT

1)Cloud Environment use to maintain the database instead of web server. n the SaaS model, cloud providers install and operate application software in the cloud and cloud users access the software from cloud clients[5]. The cloud users do not manage the cloud infrastructure and platform on which the application is running. This eliminates the need to install and run the application on the cloud user's own computers simplifying maintenance and support.

What makes a cloud application different from other applications is its scalability. This can be achieved by cloning tasks onto multiple virtual machines at run-time to meet the changing work demand. Load balancers distribute the work over the set of virtual machines. This process is transparent to the cloud user who sees only a single access point[4]. To accommodate a large number of cloud users, cloud applications can be multitenant, that is, any machine serves more than one cloud user organization. It is common to refer to special types of cloud based application software with a similar naming convention: desktop as a service, business process as a service, test environment as a service, communication as a service.

2) Performance in cloud environment is much better than web server.A **virtual private network** (**VPN**)[7] extends a private network and the resources contained in the network across public networks like the Internet. It enables a host computer to send and receive data across shared or public networks as if it were a private network with all the functionality, security and management policies of the private network. This is done by establishing a virtual point-to-point connection through the use of dedicated connections, encryption, or a combination of the two.

3) Cloud Environment having fast processing speed & data flow is high. Setting up a cloud environment is complicated, and it involves multiple elements such as database, network infrastructure, security, etc., (depending on the need). How do you increase the performance of this environment? We start small by focusing on one element at a time and try to speed up the process. In this blog, I will share two popular cloud workloads and some key things that were done to increase their performance, followed by a list of resources that you can reference for more optimization ideas...these different approaches to increase performance, which may trigger creative ideas for what you can do with your cloud environment

III. STAGES

- User
- Faculty
- Administrator

1) User: - Group registration or individual registration is the two way to part of the process. if the group having no. of person then they add the details should include information's like group strength, area of interest, semester details. Group code will be provided and use is a Individual details like address, educational qualification including current total GPA[1]. After there is a one online test for each participant .The selection procedure is base on the out of ten grade point. Out of ten a grade point will be provided based on the online test & educational qualification.if the group is no of student, then we will consider Average grade point of team members.Based on this average, project is allocated

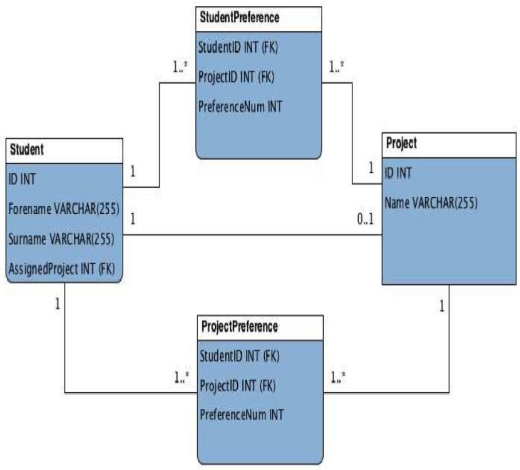

Fig 1. ER diagram for Cloud Database of Student[7]

2) Faculty:- if the user is faculty member of any institute then he register as a project guide. Maximum three project will be allocate to the one guide. he can. give the idea and moderated training to the student as per there requirement.

3) Administrator:-Admin can add, remove and edit the client project whenever he wants to take an action on it. he can manage the temporary and non -repetitive activities of client .here the student as a client[2]. Admin have a rights of introduce the right people at the right time to address the right issues and make the right decisions.

Workflow

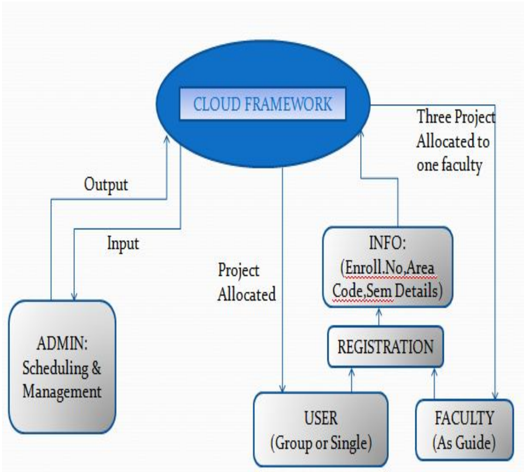

Fig 2.Framework between admin and client

IV. IMPLEMENTATION AND TESTING

Despite having agreed on using PHP as the programming language with which to develop the project allocation program, Java was initially used to develop the basic underlying algorithm, as it was the programming language most familiar to the author. With the initial Stable Marriage algorithm having been implemented, it was important to test that the algorithm performed as expected before further enhancements were made. Running both programs side by side, the matches produced by each program were identical. Making certain that this was the case, the male and female preference lists were changed on both programs whilst remaining identical to one another. After invoking the matching process for a second time, both programs returned the same results[7].

when the user either it may be single or in group of user ,firstly they want to register with server of cloud .he want to provide their basic personal information to cloud and after that he provide educational qualification and details about the institute .and proceed to next window. next window shows the one online test for each client. it is based on the out of ten grade point. The request send to cloud and cloud forward the request to the admin. admin can see the requirements of the client. and he send the report to the client. on the basis of the out of ten grade point and educational qualification ,project is allocated to the student. secondly we can interface with faculty member which is work as the guide ,also they want to provide the their basic information to the server of cloud and admin can allocated three project to the one faculty member. so that he can guide the respective student of the recognize institute[7].

V. CONCLUSION

This project was tasked with developing an automated system that would allow students, about to undertake their third year of study, to be matched with a dissertation project[6]. In doing so, the designed system was tasked with providing a reduction in the administration time needed to facilitate this process, whilst, if possible, equaling or increasing the amount of successful student project matches based on their original first choice, over that of the current system. By striving to achieve this, it is hoped that as a result, the overall level of student satisfaction surrounding this necessary process will increase[7]. This project has covered a wide area of background research into topics that are relevant to the processes that led up to the design stage of the project allocation system. We have seen how resource allocation, constraint satisfaction and conflict resolution all overlap and provide a useful framework in determining what realistic constraints can be imposed on such a system whilst still achieving a satisfactory level of success. Balancing the needs and preferences of students against needs and preference of the academics is a delicate task which will, more often than not, involve compromises having to be made in order to realize the end goal, which is in the collective benefit of both students and academics.

REFERENCE

[1]. Gamze Karayaz, Charles B. Keating, Morgan Henrie," Designing Project Management Systems" , 44th Hawaii International Conference on System Sciences@2012

[2]. Martin Lerch and Patrick Spieth," Innovation Project Portfolio Management :A Quantitative Analysis ",IEEE TRANSACTIONS ON ENGINEERING MANAGEMENT, VOL.60, NO.1@2013.

[3]. Wei-Neng Chen, Jun Zhang," Ant Colony Optimization for Software Project Scheduling and Staffing with an Event-Based Scheduler", IEEE TRANSACTIONS ON SOFTWARE ENGINEERING, VOL. 39, NO. 1,@2013

[4]. Xiaopeng Lin1, Yiyang Li1, Huaiyu Dai and Donghui Guo," Architecture of Web-EDA System Based on Cloud Computing and Application for Project Management of IC Design",@2010

[5]. Hsu-Chia Cahng, Yi-Ming Kao, Chi-Chun Lo" ,Client-Cloud Computing: A Framework for Service Integration, International Symposium on Computer, Consumer and Control,@2012

[6]. Peerasit Patanakul," Key Drivers of Effectiveness in Managing a Group of Multiple Projects", IEEE TRANSACTIONS ON ENGINEERING MANAGEMENT, VOL. 60, NO. 1@2013

[7]. Peter Wilhelm Morgan Supervisor: Mike Stannett " A Project Allocation System"@2012

HUMAN COMPUTER INTERACTION USING COMBINATION OF FACE PATTERNS AND EYE MOVEMENT (A REVIEW)

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Abstract- The human face and eye plays an important role in many aspects of verbal and non-verbal communication. Face Eye Human Computer Interaction System provides an interaction system between a computer and a human on the basis of human eye and face movement. Although gaze has been the traditional focus of eye-based HCI, eye movements provide additional information that could be useful to human-computer interfaces. The System will create patterns for action on the basis of eye and face movement and perform these actions like copy, paste, edit and so on. Module wise the project could be divided under Human Interaction by Camera, Image Processing and content Retrieval, Create Action and perform , display graph of testing phase. The algorithms for coordinates of the points on the screen that are computed by the gaze-tracking algorithm based on cross-ratio-invariant, and a procedure of calibration is needed to eliminate the error produced by the deviation of the optical and visual axes of the eyeball. The gaze tracking algorithm uses the property of cross ratio- invariant in projective transformations, and a calibration algorithm is needed. The ellipse fitting algorithm is applied to fit a standard ellipse or circle based on the coordinates of pupil edge pixels. This is a tentative algorithm it could be modified at project designing.

I. INTRODUCTION

The project title reflects the actual working and meaning of the project. The project is of human computer interaction through facial expression and eye movement combination. The problem definition is based on the application level for the paralysed person or the disabled persons who could not operate the computer by their hands or even legs. Thus a person having some facial expression as well as good eye moments could be able to handle to operate the computer using his or her facial expressions. This project could be further enhanced to use the person's neural waves to operate the system or for HCI.

Up till now the work in HCI focuses on the use of hand movements , face movements or expression ,even eye movements were used ,but in this project the combination of face and eye movements will be unique in nature. Hear, one more innovation is of not using any extra hard ware support the process of HCI will be carried out just by using a camera (inbuilt camera or web camera of high mega pixels.). This project will be totally software based project and could be used on any machine of common specifications.

Technologies used in the project are image processing and content retrieval then pattern of Action on the basis of human eye movement and face expressions then it will perform actions by making the combination of movement of eye and face expression and will make it actionable according to the commands specified in the data base. There are some predefined algorithms used in this project they are gaze-tracking algorithm based on cross-ratio-invariant, and a procedure of calibration is needed to eliminate the error produced by the deviation of the optical and visual axes of the eyeball. The gaze tracking algorithm uses the property of cross ratio-invariant in projective transformations, and a calibration algorithm is needed.

II. THE MODEL OF THE SCHEME

The model of the scheme is introduced from three aspects: the hardware of the scheme, video processing and the gaze tracking algorithm,
Each part will be discussed in the following sections.

The eye-moving video is captured under the hardware condition concerned above, and the frame rate of the video is 32 frames per second. The centre of the pupil and the corneal reflexes are detected by processing the video frame.

B. Video Processing
The video is processed in four steps. First, the image of the eye region is detected. Then the pupil image will be extracted from the image of the eye region. Next, we will detect the four corneal reflexes and calculate the coordinates of the reflexes in the video frame. Finally, the coordinates of the pupil centre points in the frame will be determined.

III. IMAGE PROCESSING

Image processing is the art and science of manipulating digital images. It stands with one foot firmly in mathematics and the other in aesthetics, and is a critical component of graphical computer systems. Image processing is any form of signal processing for which the input is an image, such as photographs or frames of video; the output of image processing characteristics or parameters related to the image. It is sometimes of interest to process a single sub region of an image, leaving other regions unchanged. This is commonly referred to as region of- interest (ROI) processing. Here we show an example of edge detection in a triangular ROI using a real image.

1) Eye Detection: The image of the eye region is detected by searching the difference among the frames during the blink motion. Natural blinking movement of both eyes is required. Let $It(x, y)$ is the current video frame image with blinking, while $It-1(x, y)$ is the previous video frame image without blinking, and then the difference image will be obtained as following:

$M(x, y) = |It(x, y) - It-1(x, y)|$ (1)

Pixels of the difference image are scanned, and the binaries image $E(x, y)$ of the difference image $M(x, y)$ is $E(x, y)$ = _1, $M(x, y) \geq$ threshold 0, $M(x, y) <$ threshold (2)

The threshold is determined by an iterative process based on the image. Noise is reduced by the process of erosion. The positions of the left and the right eye are obviously in the binary image, and the images of the eye region are detected.

2) Pupil Detection*:* Normally, the pixels of the pupil have a lower pixel value than others; moreover, the shape of the pupil is a round plate. The image of the eye region is binaries based on the difference of the gray level between the pupil area and the iris area, and the rough position of the pupil will be detected. The binaries image is smoothed with the purpose of reducing the noise. Next, the exact position of the pupil is determined by the vertical integral projection and the horizontal integral projection.

3) Corneal Reflexes Detection*:* To apply the gaze tracking algorithm based on the property of cross-ratio-invariant, the coordinates of the reflection points on the cornea must be known. The image of the pupil area obtained above mainly contains two parts: the intact pupil and part of the iris. Among all the pixels in this image, the gray level of the reflection points is the highest, moreover, the number of the reflection points is four and their location relationship is a quadrangle. All these characteristics above will help us to detect the reflection points.

C. Gaze Tracking

The gaze tracking algorithm uses the property of cross ratio- invariant in projective transformations, and a calibration algorithm is needed. The gaze tracking algorithm and the calibration algorithm will be introduced respectively.

1) Gaze Tracking Algorithm Used Cross Ratio: This gaze tracking algorithm was introduced by D. H. Yoo in 2001 [4]. The algorithm uses the property of cross-ratio-invariant in projective transformations. Two planes are concerned in the algorithm: the plane of the monitor and the plane of the image captured by the camera. The corresponding cross ratios on the two planes are equal, and the coordinates of the gaze points on the monitor will be determined.

Fig.

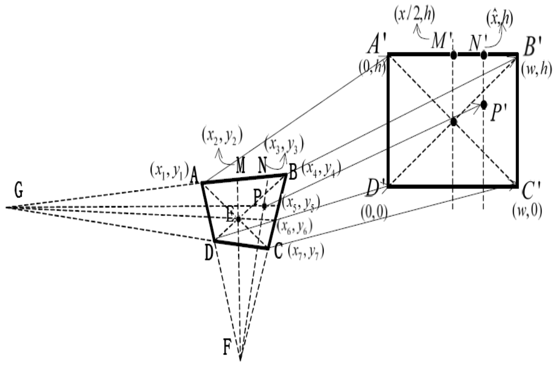


Fig. 1. The relationship between the plane of the monitor and the plane of the frame image.

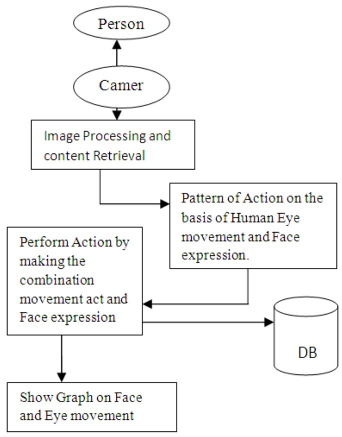
Fig.1. shows the relationship between the plane of the monitor and the plane of the frame image. *A_,B_,C_,D_* are four near infrared light sources on the corners of the monitor, and *P_* is the point the user looking at on the monitor. Correspondingly, *A, B,C,D,P* are the reflection points of the four near infrared light sources and the centre of the pupil. That is to say, *A, B, C, D, P* are the projections of *A_,B_,C_,D_,P_* on the frame image. The ordinates of *A, B, C,D* and *P* have been obtained by the previous steps, while the coordinates of *A_,B_,C_,D_* are measurable. To track gaze, the coordinate of the gaze points *P_* have to be 52 computed. The property of cross ratio invariant is used to compute it.

New technology used in this project: Hear the main focus is on facial expression and eye movement, thus we are using the combination result of both these technologies.

System Architecture: Fig 2: Face Eye Human Interaction System.

Get Face Image

Send captured Image



The project is divided into certain modules of working which includes human interaction by camera, image processing and content Retrieval, create action and perform, show graph from database images.

IV. ADVANTAGES

This project could be used by the paralysed person who are unable to move hands or legs, but are having face and eye movement in a good manner.

V. FUTURE SCOPE

The project could bring a drastic change in the world of Human computer interaction as we could also use neural technology for the interaction purposes.

REFERENCES

[1] C.S. Lin, H.T. Chen, C.H. Lin, M.S. Yeh, and S.L. Lin. Polar coordinate mapping method for an improved infrared eye-tracking system. Journal of Biomedical Engineering-Applications, Basis and Communications, vol.17, no.3, pp.141-146, 2005.

[2] C.H. Morimoto, D. Koons, A. Amit, M. Flickner, and S. Zhai. Keeping an eye for HCI. Computer Graphics and Image Processing, pp.171 -176, 1999.

[3] T. Ohno, N. Mukawa, and A. Yoshikawa. Gaze tracking system for gazebased human-computer interaction. NTT Technical Review, vol.1,no.2, pp.35-39, 2003.

[4] D. H. Yoo, J. H. Kim, B. R. Lee, and M. J. Chung. Non-contact eye gaze tracking system by mapping of corneal reflections. Fifth IEEE International Conference on Automatic Face and Gesture Recognition, pp.94 - 99, 2002.

[5] D. H. Yoo and M. J. Chung. A novel nonintrusive eye gaze estimation using cross-ratio under large head motion. Comput. Vis. Image Understand, Special Issue on Eye Detection and Tracking, vol.98, no.1, pp.25-51, 2005.

[6] D. Beymer and M. Flickner. Eye gaze tracking using an active stereo head. Proceedings of the IEEE Computer Society Conference on Computer Vision and Pattern Recognition(CVPR ’03), vol.2, pp.451-458, 2003.

[7] Xiaohui Yang, Jiande Sun, Ju Liu, Jinyu Chu, Wei Liu and Yuling Gao . A Gaze Tracking Scheme for Eye-Based Intelligent Control . Proceedings of the 8th World Congress on Intelligent Control and Automation July 6-9 2010, Jinan, China.

FULL-BRIDGE DC–DC CONVERTER WITH A FLYBACK SNUBBER FOR HYBRID APPLICATIONS

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Abstract- An isolated bi - directional full bridge DC - DC converter with flyback snubber for supplying a resistive load is simulated and experimentally verified. The DC-DC converter for high conversion ratio, high output power and soft start- up capability is presented in this paper. The circuit consist of a capacitor, a diode and a flyback converter. These components help to clamp the voltage spikes caused by the current difference between the current fed inductor and leakage inductance of the isolation transformer. The switches are operated by soft- switching technology. The suppression of inrush current which is usually found in the boost mode start up transition is presented here. The simulated and experimental results for output voltage, output current and power for both buck and boost mode are presented.

Keywords- *Isolated bidirectional full bridge converter, flyback snubber, flyback converter, soft- start up, embedded control.*

I. INTRODUCTION

A DC- to- DC converter converts a source of direct current (DC) from one voltage level to another. These converters are important in portable electronic devices such as cellular phones and laptop computers, which are supplied with power from batteries primarily. The ordinary circuit of a DC- DC converter for high power applications typically includes a bidirectional full bridge DC- DC converter [1]. The current difference between the inductor and isolation transformer does not ensure a well-defined output voltage and is characterized by less reliability and efficiency. The output voltage contains voltage spikes. An active commutation can be used to control the current in the leakage inductance. But it requires an additional clamping circuit to suppress the voltage spikes. An RCD passive snubber can be used to clamp the voltage. A buck converter was employed to replace RCD snubber. But it still needed complex clamping circuit. Active clamping increases the current stress on switches. Soft switching capability can be used, but it is not suitable for step down operation.

In this scheme of DC-DC converter with flyback snubber, the snubber recycles the absorbed energy in the clamping capacitor. The voltage of the clamping capacitor can be regulated by operating the flyback snubber independently. The current does not circulate through the full bridge switches and hence the current stress can be reduced, improving the system reliability significantly [1].

II. BASIC TOPOLOGY OF FLY-BACK CONVERTER

Fig. 1 shows the basic topology of a flyback circuit. Input to the circuit may be unregulated dc voltage derived from the utility ac supply after rectification and some filtering [7]. The ripple in dc voltage waveform is generally of low frequency and the overall ripple voltage wave form repeats at twice the ac mains frequency. Since the SMPS circuit is operated at much higher frequency (in the range of 100 kHz) the input voltage, in spite of being unregulated, may be considered to have a constant magnitude during any high frequency cycle. A fast switching device like a MOSFET, is used with fast dynamic control over switch duty ratio (ratio of N time to switching time-period) to maintain the desired output voltage.

The transformer is used for voltage isolation as ell as for better matching between input and output voltage and current requirements. Primary and secondary windings of the transformer are wound to have good coupling so that they are linked by nearly same magnetic flux. In a normal transformer, under load, primary and secondary windings con-duct simultaneously such that the ampere turns of primary winding is nearly balanced by the opposing ampere-turns of the secondary winding (the small difference in ampere-turns is required to establish flux in the non-ideal core). Since primary and secondary windings of the y-back transformer don't conduct simultaneously they are more like two magnetically coupled inductors and it may be more appropriate to call the fly-back transformer as inductor-transformer. The output section of the y-back transformer, which consists of voltage rectification and filtering, is considerably simpler than in most other switched mode power supply circuits. As can be seen from the circuit Fig. 1, the secondary winding voltage is rectified and filtered using just a diode and a capacitor. Voltage across this filter capacitor is the SMPS output voltage, It may be noted here that the circuit shown in fig. 1 is rather schematic in nature.

Amore practical circuit will have provisions for output voltage and current feedback and a controller for modulating the duty ratio of the switch. It is quite common to have multiple secondary windings for generating multiple isolated voltages. One of the secondary outputs may be dedicated for estimating the load voltage as well as for supplying the control power to the circuit. Further, as will be discussed later, a snubber circuit will be required to dissipate the energy stored in the leakage inductance of the primary winding when switch 'S' is turned off.

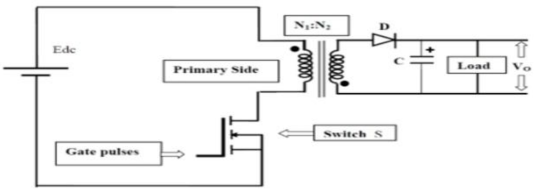

Fig1. Fly Back Converter

III. CONFIGURATION & OPERATION

The proposed isolated bidirectional full-bridge dc–dc converter with a flyback snubber is shown in Fig.2 .The converter is operated in two modes: buck mode and boost mode. Fig.2 consists of a current-fed switch bridge, a fly back snubber at the low-voltage side, and a voltage-fed bridge at the high-voltage side. Inductor Lm performs output filtering when power flows from the high-voltage side to the batteries, which is denoted as a buck mode. On the other hand, it works in boost mode when power is transferred from the batteries to the high-voltage side. Furthermore, clamp branch capacitor CC and diode DC are used to absorb the current difference between current-fed inductor Lm and leakage inductance Lll and Llh of isolation transformer Tx during switching commutation [1] [2]. The fly back snubber can be independently controlled to regulate VC to the desired value, which is just slightly higher than VAB. Thus, the voltage stress of switches M1–M4 can be eliminated to a low level. The major merits of the proposed converter configuration include no spike current circulating through the power switches and clamping the voltage across switches M1–M4, improving system reliability significantly. Note that high spike current can result in charge migration, over current density, and extra magnetic force, which will deteriorate in MOSFET carrier density, channel width, and wire bonding and, in turn, increase its conduction resistance.

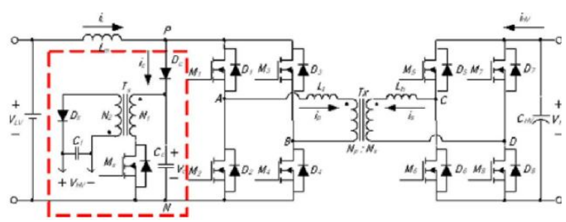

Fig.2. Isolated bidirectional full-bridge dc–dc converter with a flyback snubber

A bidirectional dc–dc converter has two types of conversions: step-up conversion (boost mode) and step-down conversion (buck mode). In boost mode, switches M1–M4 are controlled, and the body diodes of switches M5–M8 are used as a rectifier. In buck mode, switches M5–M8 are controlled, and the body diodes of switches M1–M4 operate as a rectifier. To simplify the steady-state analysis, several assumptions are made, which are as follows.

1. All components are ideal. The transformer is treated as an ideal transformer associated with leakage inductance.
2. Inductor Lm is large enough to keep current iL constant over a switching period.
3. Clamping capacitor CC is much larger than parasitic capacitance of switches M1–M8.

A. Step-up Conversion

In boost mode, switches M1 ~M4 are operated like a boost converter, where switch pairs(M1, M2) and(M3, M4) are turned on to store energy in Lm. At the high-voltage side, the body diodes of switches M5 ~ M8 will conduct to transfer power to VHV. When switch pair(M1, M2) or(M3, M4) is switched to(M1, M4) or(M2, M3), the current difference iC(=iL – ip) will charge capacitor CC and then raise ip up to iL. The average power PC transferred to CC can be determined as follows:

$$Pc = \frac{1}{2}Cc\ [(iLZo)^2 + 2iLZoV_{C(R)}]f_s$$

$$Z_0 = \sqrt{\frac{Leq}{Cc}}$$

VC(R) stands for a regulated VC voltage which is close to (VHV ·NP /NS), fs is the switching frequency and Lm>>Leq. Processed power PC will be transferred to the high-side voltage source through the flyback snubber, and the snubber will regulate clamping capacitor voltage VC to VC(R) within one switching cycle Ts(= 1/ fs). Note that the flyback snubber does not operate over the interval of inductance current ip increasing towards iL. The processed power PC by the flyback snubber is typically around 5 % of the full load power for low-voltage applications. With the flyback snubber, the energy absorbed in CC will not flow through switches M1 ~ M4,which can reduce their current stress dramatically when Leq is significant. Theoretically, it can reduce the current stress from 2iLto iL.

The operation waveforms of step-up conversion are shown in Fig. 3. A detailed description of a half-switching cycle operation is shown as follows.

Mode 1: The equivalent circuit is shown in Fig.3a. All the four switches M1 & M4 are turned on. In this interval, the inductor Lm is charged by and the current iL increases linearly. The primary windings of the transformer are short circuited.

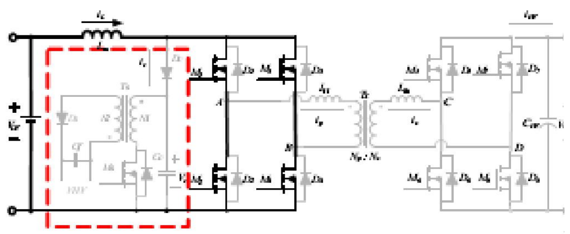

Fig.3 a. Mode1

Mode 2: The equivalent circuit is shown in Fig.3b. M1 & M4 are conducting while M2 & M3 are turned off. Clamping diode Dc conducts until the current difference drops to zero. and D5 & D8 conducts to transfer power. The current difference flows into clamping capacitor Cc.

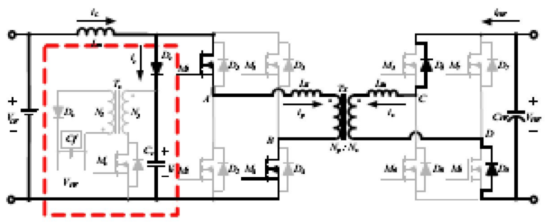

Fig. 3b. Mode 2

Mode 3: The equivalent circuit is shown in Fig.3c. Now Dc stops conducting and flyback snubber starts to operate. Clamping capacitor Cc is discharging and flyback conductor stores energy. M1& M4 in ON state and D5 & D8 remain ON to transfer power.

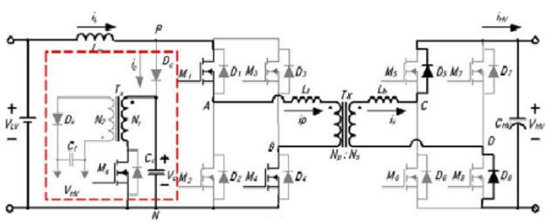

Fig.3c. Mode 3

Mode 4: The equivalent circuit is shown in Fig.3d. At, the energy stored in flyback conductor is transferred to high voltage side. The flyback snubber operates to regulate VLV to VC to VC(R). Same switches operate to transfer power from VLV to VHV.

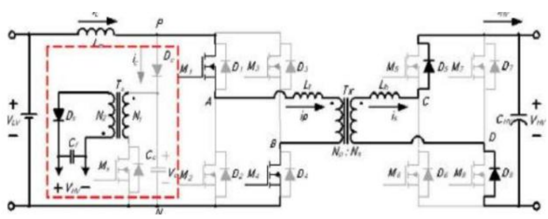

Fig.3d. Mode 4

Mode 5: The equivalent circuit is shown in Fig.3e. At, we obtain a regulated voltage VC(R). The main power stage is still transferring power from VLV to VHV. It stops at and a half switching cycle operation is completed.

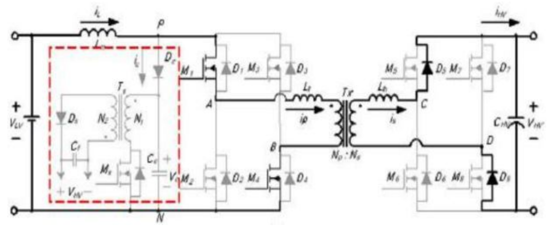

Fig.3.e. Mode 5

B. Step-down Conversion

In the analysis, leakage inductance of the transformer at the low-voltage side is reflected to the high-voltage side. This circuit is known as a phase-shift Full-bridge converter. In the step-down conversion, switches M5–M8 are operated like a buck converter, in which switch pairs (M5,M8) and (M6,M7) are alternately turned ON to transfer power from VHV to VLV. Switches M1–M4 are operated with synchronous switching to reduce conduction loss. For alleviating leakage inductance effect on voltage spike, switches M5–M8 are operated with phase-shift manner.

Mode 1: The equivalent circuit is shown in Fig.4a. Now M5 & M8 are turned ON. The VHV is immediately excited on the transformer and the whole voltage is excited on Leq. The transformer current increases linearly towards the load current. M1& M4 conducts to transfer power.

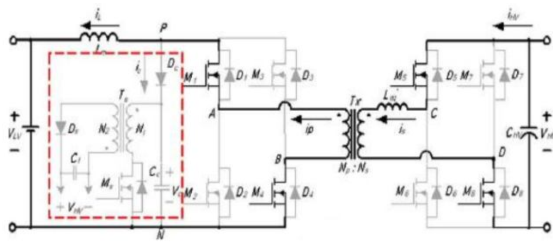

Fig.4.a. Mode 1

Mode 2: The equivalent circuit is shown in Fig.4b. M8 remains conducting while M5 is turned OFF. D6 conducts the free wheeling leakage current. The transformer current reaches the load current level. DC conducts the resonant Leq and the clamping capacitor CC.

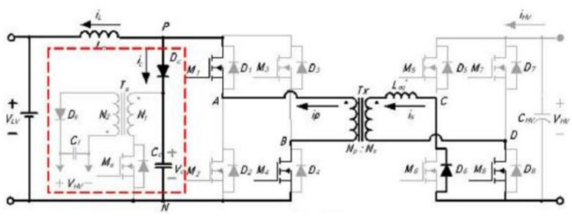

Fig.4.b. Mode 2

Mode 3: The equivalent circuit is shown in Fig.4c. With diode D6 conducting, M6 is turned.

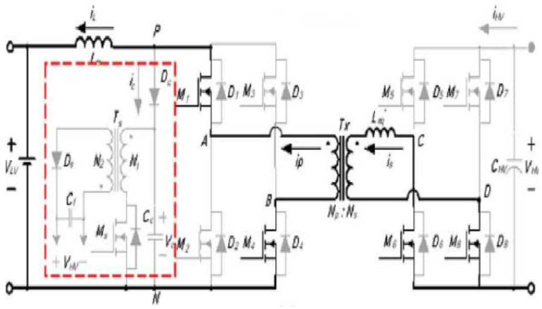

Fig.4c. Mode 3

Mode 4: The equivalent circuit is shown in Fig.4d. M6 remains conducting while M8 is turned OFF. The diode D7 starts to conduct the freewheeling leakage current.

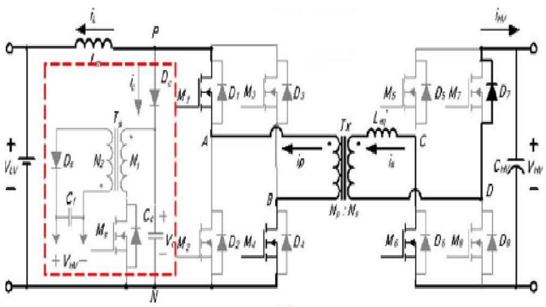

Fig. 4.d. Mode 4

Mode 5: The equivalent circuit is shown in Fig.4e. With conducting, M7 in turned ON with ZVS. Over this interval, the active switches change to the other pair of diagonal switches and the voltage and the transformer reverses it polarity. It stops now and it completes a half switching cycle operation [1][2].

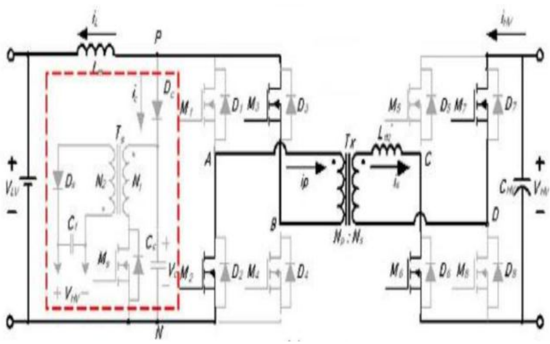

Fig. 4.e. Mode 5

IV. SIMULATION RESULTS

For comparison, three prototypes, the dual full-bridge converters with an RCD passive snubber, an active clamping circuit, and the proposed flyback snubber, were simulated. An input voltage of 48 V is applied and an output of 360 V is obtained for boost topology and reverse for buck topology. Closed loop simulation is done for getting the following results.

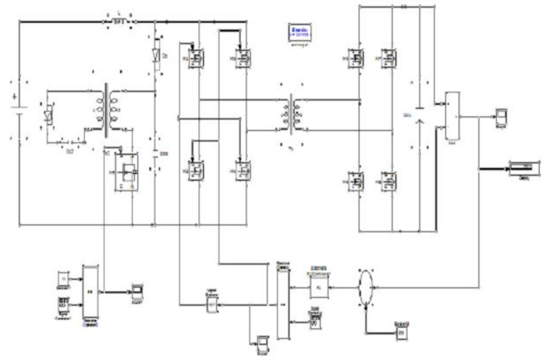

Fig.5.a. Closed loop Boost topology

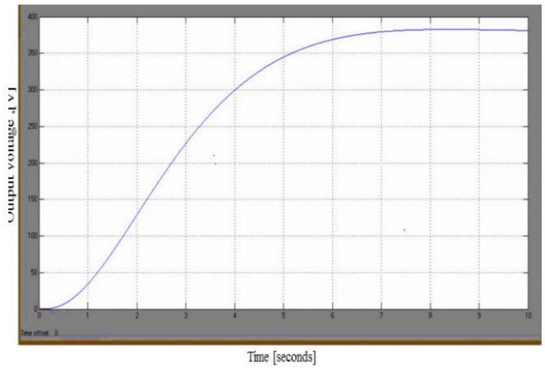

Fig.5.b. Simulation result of closed loop boost Topology

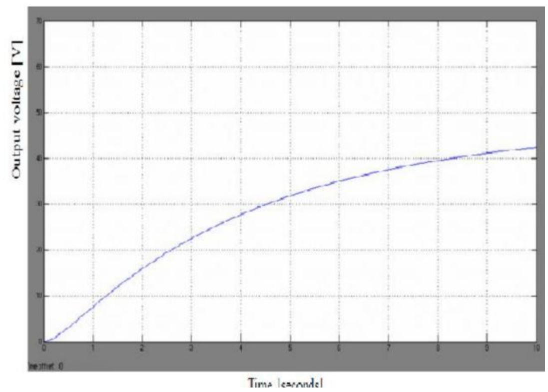

Fig.5.c. Simulation result of closed loop buck topology

V. CONCLUSION

An isolated bi-driectional full bridge DC-DC converter transformer converter had voltage spike due to the current difference between the current fed inductor and leakage inductance of the isolation transformer. This voltage spike has been alleviated by the fly back snubber. The fly back snubber can be controlled to attain a soft start-up feature. The current stress is reduced under heavy load conditions. This converter also has the advantage of increased reliability and efficiency.

REFERENCE

[1] Tsai-Fu Wu, Yung-Chu Chen, Jeng-Gung Yang, and Chia-Ling Kuo, "Isolated Bidirectional Full-Bridge DC–DC Converter with a Flyback Snubber," in IEEE transactions on power electronics, vol. 25, no. 7, july 2010.

[2] T .-F . Wu, Y .-C. Chen, J.-G . Y ang, Y .-C. Huang, S.-S. Shyu and C.-L. Lee, "1.5 kW Isolated Bi-directional DC-DC Converter with a Flyback Snubber," .

[3] Hamid R. Karshenas, Hamid Daneshpajooh, Alireza Safaee, Praveen Jain and Alireza Bakhshai "Bidirectional DC-DC Converters for Energy Storage Systems"

[4] Shahab H. A. Moghaddam, Ahmad Ayatollahi, Abdolreza Rahmati "Modeling and Current Programmed Control of a Bidirectional Full Bridge DC-DC Converter"

[5] K. Banu priya, JBV Subrahmanyam, Ch.Srikanth & PK Sahoo Bandana "Isolated bidirectional full-bridge dc–dc converter with fly back snubber for high-power applications".

[6] D. Kirubakaran and Rama Reddy Sathi, "Embedded Controlled Isolated Bidirectional Full-Bridge DC-DCConverter with Flyback Snubber,"

[7] "DC to DC Converters" Version 2 EE IIT, Kharagpur.

AN IMPROVED METHOD FOR POWER FACTOR IMPROVEMENT OF ELECTRONIC BALLAST FOR MULTIPLE FLUORESCNT LAMPS

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Abstract- High-power-factor (HPF) electronic ballast with boost topology for multiple (two) fluorescent lamps are presented in this paper. The simulation is done for two 36W fluorescent lamps in MATLAB. The input is 110V single phase AC supply at 60 Hz. The circuit comprises of a boost converter which acts as an inherent power factor corrector (PFC). Single stage topology has been taken into consideration which reduces the requirement of extra power switches. By integrating the power switches used in boost converter and half bridge inverter, new single stage HPF electronic ballast is implemented which is used to both correct the input power factor and drive the fluorescent lamp. Performance analysis and simulation results prove that improved power factor can be obtained along with improved ballast efficiency.

Index Terms- *Electronic ballast, discontinuous conduction mode(DCM), single stage boost topology, switching frequency, power factor corrector (PFC), high-power-factor (HPF),*

I. INTRODUCTION

Lighting ballast is a piece of equipment required to control the starting and operating voltages of electrical gas discharge lights. The term lighting ballast can refer to any component of the circuit intended to limit the flow of current through the light, from a single resistor to more complex devices. Ballast is used to perform the following two functions:

• Provide the starting kick.

• Limit the current to the proper value for the tube you are using.

The basic block diagram of proposed PFC circuit with high frequency electronic ballast is shown in Fig.1. The PFC stage consist of an active power switch Q, a diode D, an inductor Lb and a dc-link capacitor Cdc. When the active power switch Q is turned on,the inductor Lb draws current from the ac line source.The active power switch Q is switched on and switched off at a high frequency, so the input current also becomes a pulsating waveform at the same frequency. By properly controlling the amplitude and duration of the pulsating current, the average of the input current can be made to be sinusoidal and in phase with the ac input voltage source. For controlling purpose we are using a control circuit. Consequently, a nearly unity PF and very low THD can be achieved.

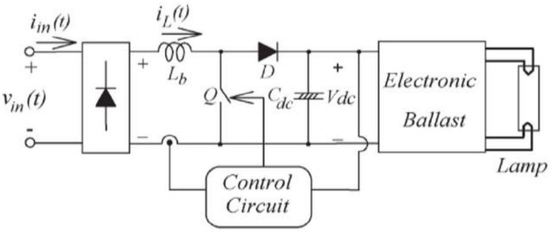

Figure 1: Basic block diagram of proposed PFC Circuit

From fig.1, Vm is the peak value of input voltage and Vdc is output voltage. The following equation shows relation between Vm and Vdc.

$$\propto = \frac{Vm}{Vdc}$$

To get a unity power factor value of α must be minimum. From the above equation it is clear that, the value of α can be controlled by controlling the values of peak input voltage, Vm and output voltage Vdc, but peak input voltage, Vm is constant. So in order to control the value of α we must control Vdc. maximum value of Vdc means minimum value of α.The control circuit senses output voltage and compares it to an internal reference DC voltage error amplifier , and sets on time of Q to keep output voltage to a selected value. The second task of power factor correction circuit is to sense the input current and force it to have a sinusoidal wave shape in phase with the input line voltage.[1][6]

II. CIRCUIT CONFIGURATION

Fig.2 shows the circuit configuration for proposed high power factor electronic ballast. It consists of two active power switches Q1 and Q2.The boost converter is operating in the discontinuous conduction mode.

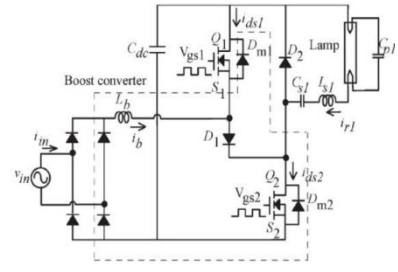

Fig.2. Proposed ballast for one fluorescent lamp.

The circuit for one fluorescent lamp can be extended to use for 2,3 or 4or any number of fluorescent lamps in parallel. And the circuit configuration for 2 fluorescent lamp is shown below. From the figure it is clear that the proposed ballast requires lesser number of active power switches than the traditional electronic ballast.

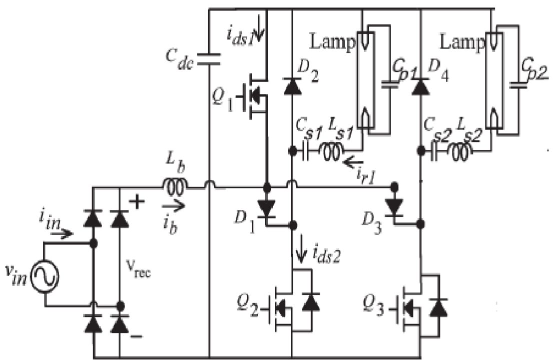

Fig.3.Proposed ballast for two fluorescent lamps

III. OPERATION

The active power switches, Q1 and Q2, are excited by two complementary signals, Vgs1and Vgs2, with a short dead time. Neglecting the dead time, the duty cycle of Vg1 is(1 - D) when that of Vg2 is D. To ensure zero-current switching of Q1 and ZVS of Q2 at the instant when they are turned on, the switching frequency of the series - resonanance parallel-loaded inverter should be greater than the resonant frequency of the inverter stage, so that the load circuit presents inductively with a resonant current lagging behind the fundamental component of the inverter output voltage. Meanwhile, the quality factor of the inverter stage is assumed to be high enough to have a nearly sinusoidal load current. In one switching cycle, the electronic ballast operation of the single-stage topology for multiple fluorescent lamps can be divided into six modes as follows[1].

i. Mode I

Active power switch Q2 is turned off before time 't0'. Since the load current ir is negative, freewheeling diode Dm2 conducts. The load resonant current ir flows through the freewheeling diode Dm2 and dc-link capacitor Cdc. A turn-on signal is applied to the gate of the active power switch Q2 at the beginning of this mode. .

The line voltage is imposed on inductor Lb as soon as active power switch Q2 is turned on. During DCM operation, the inductor current ib of the boost converter increases linearly from zero. Hence, the turn-on of the switch Q2 occurs at zero-current switching condition. The slope of ib is proportional to the input line voltage. In the interval of this mode, the input current Iin is equal to ib. The current of ids2 is the difference between the inductor current ib and the load resonant current ir. When the difference between ib and ir becomes positive, the diode Dm2 is turned off and mode I ends.

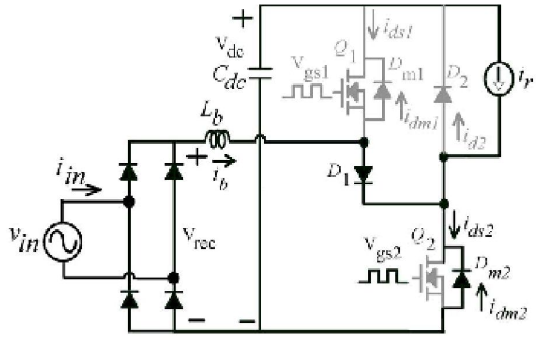

Fig.4. Equivalent circuit of mode 1

ii. Mode II

At the beginning of this mode the power switch Q2 is in the on state. Lb is continuously under the effect of line voltage and ib increases. In this mode, the currents ib and ir naturally shifts itself from diode Dm2 to the active power switch Q2. The load resonant current ir goes through the active power switch Q2 and dc-link capacitor Cdc where as both the currents ir and ib pass through the active power switch Q2.a Thus two paths are followed. One, from the line source through the inductor Lb and power switch Q2 and back to the rectifier stage constitute the boost converter circuit. Second, the resonant load current flowing through the power switch Q2 and the discharge capacitor Cdc. When the active power switch Q2 is turned off, Mode II ends and the operation enters Mode III[1][3].

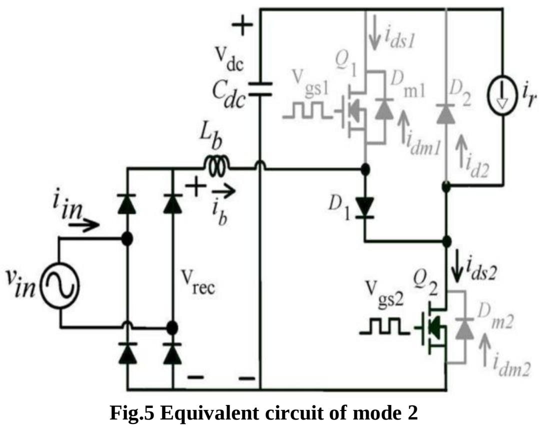

Fig.5 Equivalent circuit of mode 2

iii. Mode III

Figure 6. shows equivalent circuit of Mode III. When the gate signal Vg1 is applied the power switch Q1 comes into action. This marks the beginning of mode III. At this point of time, the inductor current ib reaches its peak value and the active power switch Q2 is turned off. The inductor current ib freewheels through Dm1 to charge the dc-link capacitor Cdc.

The load resonant current ir flows through the freewheeling diode D2.Thus two current paths can be seen. One, the load resonant current freewheeling through diode D2. Second, the inductor current charging the dc link capacitor Cdc through the diode Dm1. The voltage across Lb is equal to Vrec – Vdc. Therefore, the inductor current ib decreases linearly. When the load current ir reaches zero, mode III ends and operation enters mode IV.

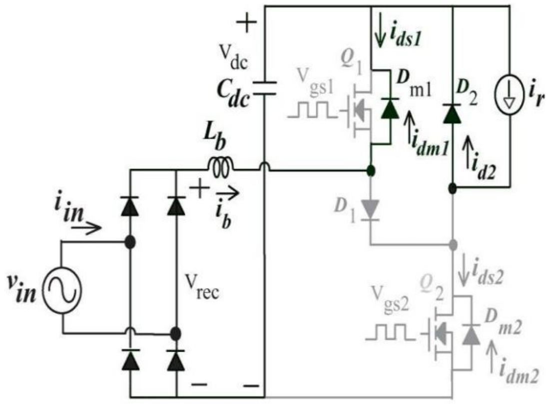

Fig.6. Equivalent circuit of mode 3

iv. Mode IV

In this mode the inductor current ib flows through diode Dm1 and charges the dc-link capacitor Cdc. The inductor current ib decreases continuously. During this mode, the load current ir goes to negative and flows through diodes D1 and Dm1. Mode IV finishes at the time when the inductor current |ib| Equals load current |ir|, and then, the operating mode enters MODE V. At this instant, the current |ir| − |ib| naturally shifts from the diode Dm1 to the active power switch Q1. That is to say, the active power switch Q1turns on softly at the zero-current-switching condition to reduce the switching losses. Figure 7 shows equivalent circuit of Mode IV [1][3].

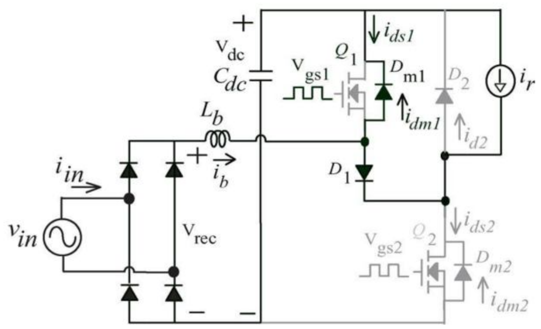

Fig.7. Equivalent circuit of mode 4

v. Mode V

The active power switch Q1 turns on at the beginning of mode V and carries both the inductor current ib and the load current ir. The load current ir goes through the active power switch Q1 and diode D1. The inductor current flows back through the active power switch Q1, dc-link capacitor Cdc, and rectifier to the ac line source. Mode V ends when the inductor current ib declines to zero. At this instant, the circuit operation enters mode VI. Fig.8. shows equivalent circuit of Mode V.

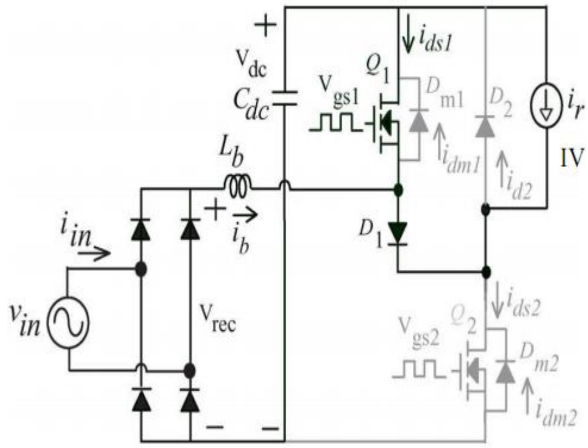

Fig.8. Equivalent circuit of mode 5

vi. Mode VI

During this operating mode, only a negative load current ir flows through the active power switch Q1 and the diodeD1. Mode VI ends when the gate signal Vg1 is applied marking the beginning of mode I of the next cycle. Figure 9 shows equivalent circuit of Mode VI.

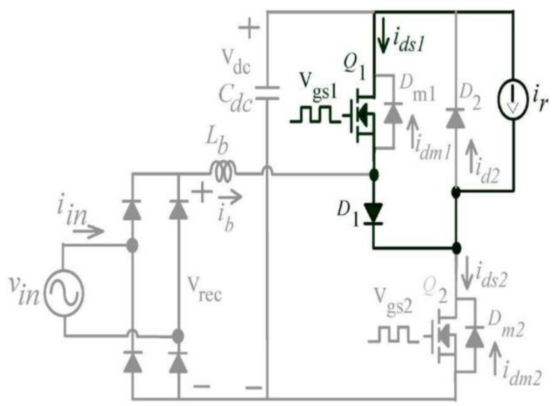

Fig.9. Equivalent circuit of mode 6

IV. WAVEFORMS

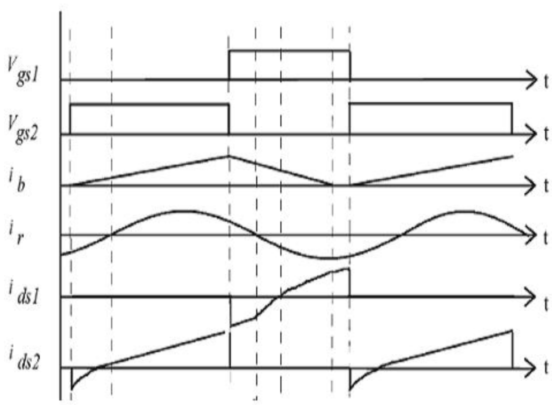

Fig.10.Waveforms of proposed electronic ballast

Figure 10 shows the current waveforms for the proposed electronic ballast.Vgs1 and Vgs2 are gate signals for MOSFETs. Ib is the inductor current. ir is the load resonant current[1][3].

V. SIMULATION RESULTS AND DISCUSSIONS

By using MATLAB/ Simulink ,the simulation of the proposed electronic ballast is done. The results obtained are given below. First the simulation is done for single stage high power factor electronic ballast for one fluorescent lamp. Then the simulation is done for stage high power factor electronic ballast for two fluorescent lamp. The parameters used for simulation are given in the following table.

Table 1: Simulation parameters

PARAMETERS	VALUE
Input voltage	110V, 60Hz
Switching Frequency	50KHz
DC link capacitor	C_{dc}=155μF
Boost inductor	L_b=0.4mH
Capacitor	C_p=15nF
Inductor	L_s=1.81mH
Capacitor	C_s=0.15μF

Simulation is done with the above given parameters and the results obtained for one fluorescent lamp is given below. Figure 11 and 12 shows the input and output waveforms for one fluorescent lamp. The input voltage and current are almost in phase, thus a power factor of 0.995 is obtained.

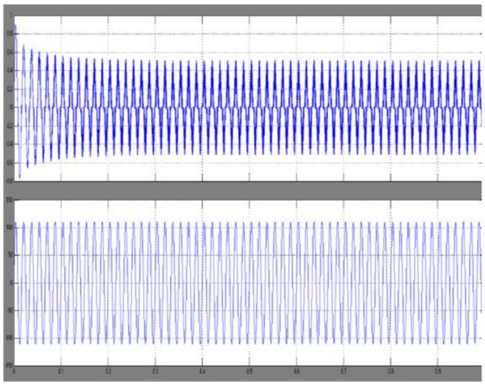

Fig.11.Input current and voltage waveform for one fluorescent lamp

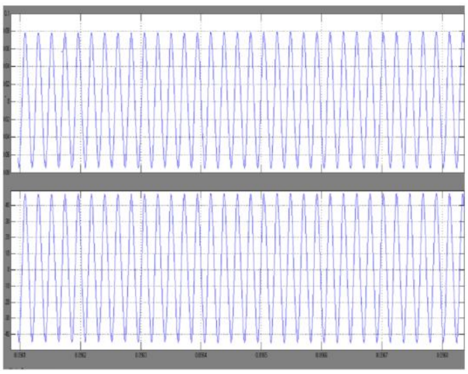

Fig.12 Output current and voltage waveform for one fluorescent lamp.

Simulation is also done for ballast for two fluorescent lamps and the results are shown in figures 14 and 15.Here also the power factor is almost unity. The Simulink model used for two fluorescent lamp lighting system is also given below. The power factor obtained is 0.997.

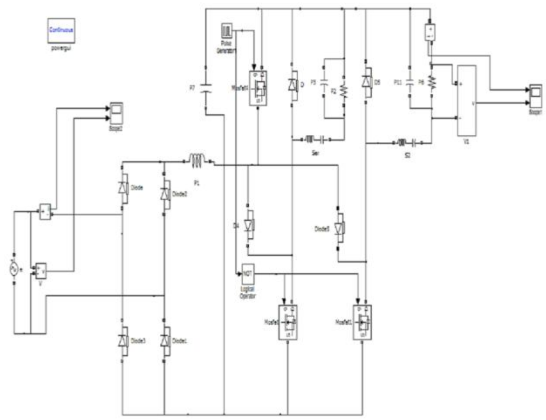

Figure 13. MATLAB /Simulink model of proposed two fluorescent lamp lighting system

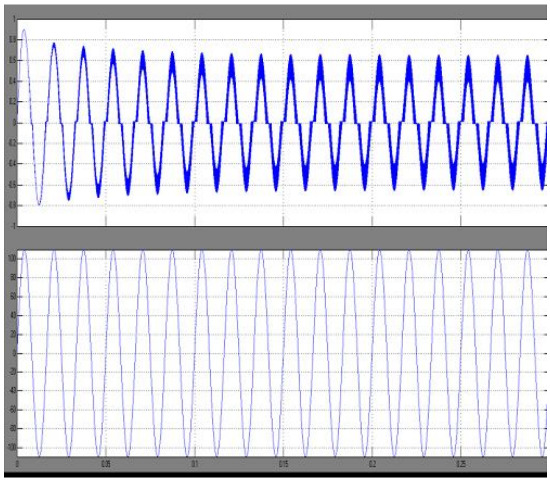

Fig.14.Input current and voltage waveforms for two fluorescent lamps.

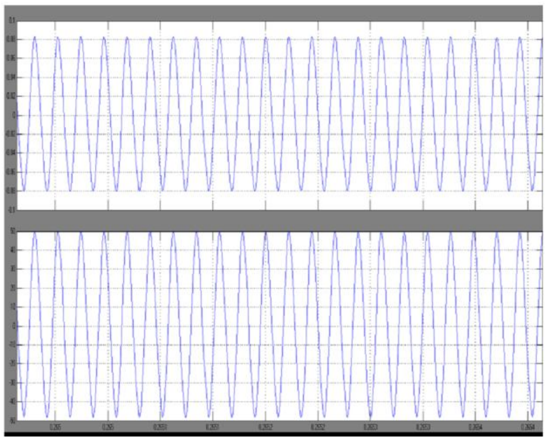

Fig.15.Output current and voltage waveforms for two fluorescent lamps

In this paper simulation is done only for one fluorescent lamp and two fluorescent lamps. It is no matter how many fluorescent lamps we are using, always the power factor is close to unity. We can extent it for more fluorescent lamps. We can use the same configuration for any number of lamps. As the number of lamps increase this configuration is more and more feasible.

VI. CONCLUSION

The paper introduces single stage electronic ballast with high power factor and low harmonic distortion for driving two 36W fluorescent lamps. The proposed electronic ballast is the cascade operation boost dc-dc converter and series-resonant parallel loaded inverter. The boost dc- dc converter acts as a power factor correction device. Simulated results have been obtained for the proposed electronic ballast. Considerable numbers of components are reduced resulting in significant reduction in cost in the proposed electronic ballast for multiple fluorescent lamps. A high power factor and reduced THD have been achieved with this electronic ballast. The proposed ballast can be used for any number of fluorescent lamps.

REFERENCE

[1] Ying chun chuang,"A novel single stage high power factor electronic ballast with boost topology for multiple fluorescent lamps",IEEE Trans. on Ind. Appl.,vol.45, no.1,pp.15,feb 2009.

[2] Yu-Lung Ke and Ying-Chun Chuang,"A novel single stage high power factor correction circuit with high frequency resonant energy tank for DC link inverters"IEEE Trans. on circuits and systems, VOL. 53, NO. 2, Feb.2006

[3] R. A. Gupta, Rohit Agarwal, Hanuman Soni and Mahankali Ajay "Design and simulation of single stage high power factor electronic ballast for multiple fluorescent lamps",International Journal of Recent Trends in Engineering, Vol 2, No. 5, November 2009

[4] Ying chun chuang and Hung liang cheng, "Single stage single switch high power factor electronic ballast for multiple fluorescent lamps "IEEET rans. OnInd. Appl. vol.43,,Dec.2007.

[5] John C. W. Lam, and Praveen K Jain,"High power factor single stage single switch electronic ballast for compact fluorescent lamps"IEEE Trans. on power electronics vol. 25, no8, Aug 2010.

[6] Abraham.I.Pressman," Switching power Supply design" McGraw-Hill, 1998.

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FREQUENCY RECONFIGURABLE ANTENNA FOR MIMO BASED WIRELESS SYSTEMS

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Abstract- With the current scenario of development of antennas in the wireless communication field, the need of compact multiband, multifunctional and cost effective antenna is on the rise. The study of reconfigurable antennas has made great progress in recent years. They are lighter in weight, smaller in dimension and lower in price. Moreover, the reconfigurable antennas can provide diversity feature of operating resonant frequency, polarization, and radiation pattern. This paper explains what the perspective of a reconfigurable antenna is in a modern communication system. It also gives the overview of the most employed switch and tuning elements that enable antenna reconfiguration. The latest researches of the frequency reconfigurable antenna are summarized in this paper to present the characteristics.

Index Terms- *reconfigurable antenna, wireless communication system, multiple input multiple output (MIMO) systems.*

I. INTRODUCTION

Wireless communication is evolving towards mutifuctionality. They are bringing information to us with convenience and flexibility at low cost. There are many wireless communication systems in our daily lives such as cellular radio systems, mobile satellite systems, wireless local area networks, etc. All these systems are still undergoing a great evolution, with constantly emerging new solutions. The necessity of diversity, and ever growing requests for the capacity of a communication channel, will soon be unsolvable by means of classical antennas. Therefore, reconfigurability will become a key need in the near future. Reconfigurable antennas can offer new capabilities for future wireless applications. With the rapid development of wireless communication system, especially the research on MIMO techniques, reconfigurable antennas is gaining great attention. MIMO wireless systems have demonstrated the potential to increase communication spectral efficiency in rich multipath environment.

Different characteristics (such as resonant frequency, radiation patterns, polarization, etc) of these novel antennas can be reconfigurable through the change of the structures. The concept of reconfigurable antenna firstly appeared in D.Schaubert's patent "Frequency-Agile, Polarization Diverse Microstrip Antenna and Frequency Scanned Arrays" in 1983. In 1999, 12 well-known universities, research institutes and companies in the United States participate in the project of "Reconfigurable Aperture Program (RECAP)", launched by the United States Defense Advanced Research Projects Agency (DARPA). By incorporating PIN diodes, varactors, radio frequency micromechanical systems (RF - MEMS), photoconductive elements or liquid crystals or by physical altering antenna structures reconfigurability of frequency, polarization ,radiation patterns or all of these can be achieved. Frequency Reconfigurable antennas have received significant attention for their applications in communications, electronic surveillance and countermeasures, by adapting their properties to achieve selectivity in frequency, bandwidth, polarization and gain.

As a result of the significance of frequency reconfigurable antennas, in this paper, the latest researches are analyzed and summarized to present the functions and the implementations of reconfigurable antennas, which are illustrated in section II and section III. In section IV, the future work and conclusion is presented.

II. FREQUENCY RECONFIGURABLE ANTENNA

Frequency reconfigurable antenna has the reconfiguration of the resonant frequency by the change of the structure, while the radiation patterns and polarization remain unchanged. So, frequency reconfigurable antenna can be applied among a very wide arrangement of frequency band or among multiple frequency bands. A lot of frequency reconfigurable antenna designs have been proposed till date. Frequency reconfigurable antennas can be classified according to their reconfiguration techniques. Lumped-elements, variable capacitors, silicon photo switches, MEMs (micro-electro-mechanical) switches or PIN diodes are usually incorporated in the design of the antenna. The characteristics of frequency reconfigurable antennas are estimated from parameters such as effective bandwidth, operating frequencies and associated applications, tunable frequency range or number of resonant frequencies, and their performance especially depend on the consistency of matching, gain and radiation pattern.

The main techniques to achieve frequency reconfiguration are, a) selectively switching in or out parts of the antenna structure, or switching between different external matching circuits; b) adjusting the loading of the antenna externally, i.e. varactor diodes; c) changing the substrate characteristic, i.e. permittivity; and d) changing the antenna geometry by mechanical movement.

Switching or tuning within an antenna or in an external circuit can be achieved by means of PIN diodes, GaAs FETs (Gallium Arsenide Field-Effect Transistor), MEMS (Micro-electro mechanical Systems) devices or varactors [1,2]. MEMS devices have the advantage of very low loss, but the disadvantages are high operating voltage, high cost and lower reliability than semiconductor devices [3]. GaAs FETs used in switching mode, with zero drain to source bias current, have low power consumption but poorer linearity and higher loss. PIN diodes can achieve low loss at low cost, but the disadvantage is that in the on state there is a forward bias dc current, which degrades the overall power efficiency. Varactor diodes have the advantage of providing continuous reactive tuning rather than switching, but suffer from poor linearity. Changing the substrate permittivity to shift the resonant frequency is another approach, but the cost may be problematic. Adjusting the resonant frequency by changing the antenna geometry using mechanical movement can provide lossless and ideal linearity.

However, it needs mechanical adjustment and requires more time for switching between frequency operating bands.

A significant number of reconfigurable antennas incorporating switches, both switching in or out parts of the antenna structure or switching between external matching circuits, have been summarized here Ref. [4] presents an antenna system consisting of two self-diplexing planar inverted F antennas (PIFAs) that are co-designed with an antenna interface module (AIM). The co-design of two PIFAs with an antenna interface module (AIM), as shown in Fig.1, allows a reconfigurable system to be achieved to cover a number of bands with reduced both dimensions and losses. By incorporating 8 switches which were connected to 8 different impedance matching circuits, 8 different operating bands were obtained, such as 0.824 to 0.849 GHz, 0.880 to 0.915 GHz, 0.925 to 0.960 GHz, 0.824 to 0.894 GHz, 1.710 to 1.785 GHz, 1.850 to 1.910 GHz, 1.850 to 1.990 GHz, and 1.920 to 2.170 GHz.

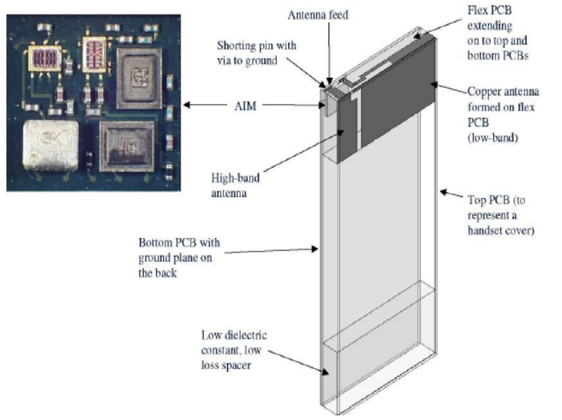

Fig 1-Structure of a five-band, seven-mode reconfigurable antenna and antenna interface module [4]

Ref. [5] describes a single port chassis antenna, which incorporates 17 matching circuits to cover from 76 MHz to 2500 MHz and 5 matching circuits to cover from 470 MHz to 2500 MHz. Switches were used to connect different impedance matching circuits. Fig. 2 shows the antenna concept i.e. chassis, of the mobile to act as antenna and a tuning network to optimize the coupling according to the requirements in the different frequency bands of the applications.

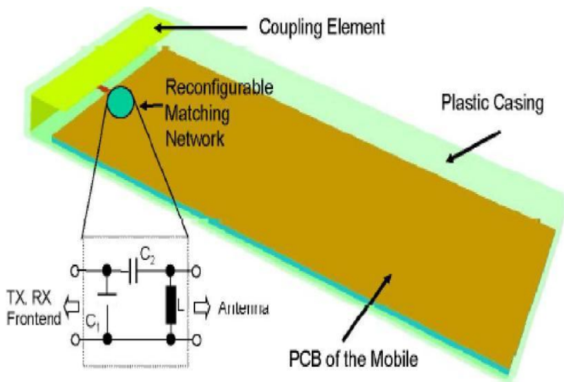

Fig. 2- Geometry of a single port chassis antenna [5]

A Vivaldi antenna, shown in Fig.3, with a capability to operate in wideband or narrowband operations is presented in ref. [6]. Four pairs of switchable ring slots were introduced to change the wideband properties into narrower pass bands. A wideband operation was achieved when all ring slots are decoupled from the tapered slot, by short circuiting all the gaps to provide a smooth flow of induced current along the tapered profile which covers from 1 to 3.2 GHz. To obtain the narrowband mode, ring slots in lowermost, middle and uppermost positions were coupled or decoupled and three sub bands, low band 1.1 GHz, mid band 2.25 GHz and high band 3.1 GHz, were achieved.

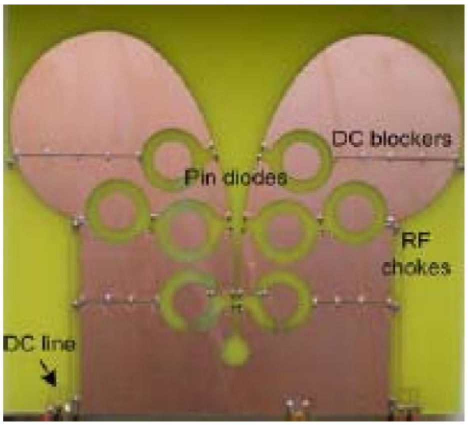

Fig.3- A Vivaldi antenna incorporating four pairs of switchable ring slots [6]

Ref. [7] proposes the use of a liquid crystal substrate for a patch antenna whose frequency can be tuned by changing the biasing voltage across the substrate, shown in Fig.4. The simulations showed that by varying the biasing voltage from 0 V through 11 V, the operating frequency of the circular patch antenna can be varied from 1.08 GHz through 2.35 GHz.

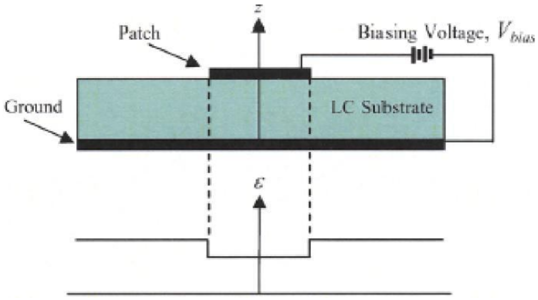

Fig. 4- A typical profile of the permittivity for a patch antenna with an LC substrate and a bias voltage. [7]

Ref. [8] presents an antenna scheme suitable for cognitive radio (CR) applications. A UWB sensing antenna and a frequency reconfigurable communicating antenna are incorporated together into the same antenna substrate, as shown in Fig. 5. Tunability was achieved by a rotational motion of a series of antenna patches. The sensing antenna can cover from 2 to 10 GHz. It is clear that each patch on the rotating section resonates at a different band and the five patches cover from 2 to 10 GHz. The radiation patterns are not quite omni-directional and become worse at higher frequencies.

Fig. 5- The fabricated prototype combined with a UWB sensing antenna and a frequency reconfigurable communicating antenna. [8]

III. MIMO ANTENNA

Designing a MIMO antenna that is able to access a very large number of multiple wireless communication standards is a formidable challenge for the mobile terminal designer, who is already short of space in which to locate the antennas. Several papers discuss MIMO antennas. Both isolation and correlation coefficient between antennas are important features. In general, lower isolation will result in lower correlation. Isolation higher than 10 dB or correlation below about 0.4 will lead to useful capacity increases in MIMO systems. The techniques to achieve low correlation are:

1. Design antenna with different radiation pattern
2. Apply decoupling network between antennas
3. Use EBG substrate to reduce mutual coupling among the antennas
4. Etch the ground plane between two antennas
5. Introduce parasitic element along the ground plane to achieve high isolation

Ref. [9] presents a combination of a monopole with dense meandering end and a PIFA with a shorted parasitic branch, as shown in fig 6. Simulation shows each of the antennas covers the bands of 869-894 MHz, 1805-2170 MHz with 6 dB return loss. The isolation is above 3 dB and 10 dB at lower and higher bands, respectively. The values of mean effective gain, at free space, for both antennas, are between -6 to -8 dBi at the lower band and - 4 to -8 dBi at higher band. It leads to the correlation, between the two antennas, dropping from 0.51 (in free space) to 0.34 (in user's hand).

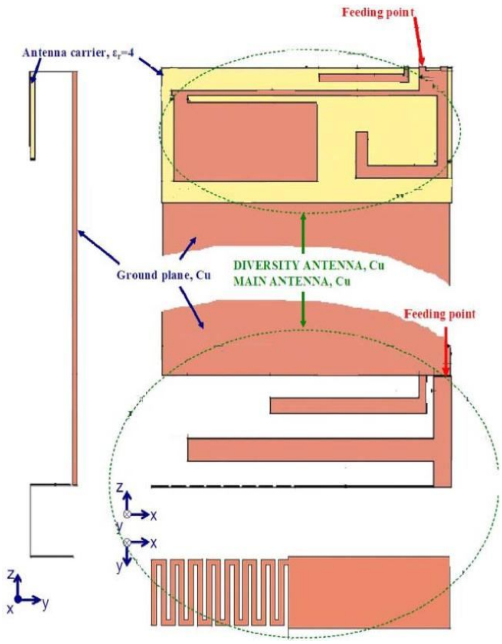

Fig. 6- Structure of a monopole with dense meandering end and a PIFA with a shorted parasitic branch [9]

Ref. [31] introduces a dual-feed planar PIFA. Two isolated feeding ports are using one common radiating plate. The main technique introduced is to etch the ground plane under the radiating plate, as shown in Fig. 7, to reduce the mutual coupling. The isolation is improved by 12 dB at 2.53 GHz.

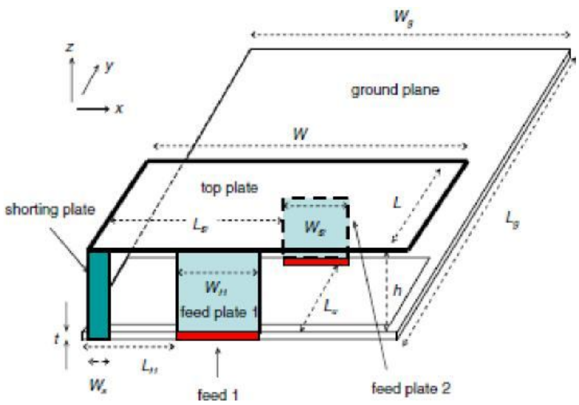

Fig. 7- Geometry of dual-feed PIFA for MIMO applications [31]

IV. CONCLUSION

This paper presents a brief review of recent research findings concerning the antennas and their design for MIMO systems. Various issues like antenna array configuration and their impact on channel capacity, selective antenna concepts, the scope of research involved in reconfigurable antennas, types of patch antennas and their usability and the main sources for mutual coupling and the ways to reduce it are discussed. From this background study, several conclusions can be drawn. Most of the designs were limited to either only single service access or narrow tuning range. In order to be able to access a very large number of multiple wireless communication standards with increased channel capacity, several MIMO antennas have been proposed to achieve low correlation using different techniques. Addressing the above issues and investigating good antennas for current and future applications are the main objective of this PhD study

V. FUTURE WORK

This paper presents a brief review of recent research findings concerning the antennas and their design for MIMO systems. Various issues like antenna array configuration and their impact on channel capacity, selective antenna concepts, the scope of research involved in reconfigurable antennas, types of patch antennas and their usability and the main sources for mutual coupling and the ways to reduce are discussed. Performances of reconfigurable antennas need to be improved for adapting to different applications. The performance of a reconfigurable antenna greatly depends on the element(s) that enable reconfiguration. Evaluation and analysis of few antennas will be done in future works. Finally, it is concluded that a lot of research is required to be done in antenna design for the better performance of MIMO systems, which form a main part for the future 4G communications.

REFEENCES

[1] Y. Yashchyshyn, "Reconfigurable antennas by RF switches technology", in Perspective Technologies and Methods in MEMS Design, 2009. MEMSTECH 2009, 5th International Conference, 2009, pp. 155-157

[2] P. Gardner, M. R. Hamid, P. S. Hall, J. Kelly, F. Ghanem, and E. Ebrahimi, "Reconfigurable antennas for cognitive radioL requirements and potential design approaches", in Wideband, Multiband Antennas and Arrays for Defence or Civil Applications, 2008 Institution of Engineering and Technology Seminar on, 2008, pp. 89-94

[3] G. M. Rebeiz and J. B. Muldavin, "RF MEMS switches and switch circuits", Microwave Magazine, IEEE, vol. 2, 2001, pp. 59-71.

[4] K. R. Boyle, P. G. Steeneken, Z. Liu, Y. Sun, A. Simin, T. Huang, E. Spits, O, Kuijken, T. Roedle, and F. Van Straten, "Reconfigurable antennas for SDR and cognitive radio", Antennas and Propag., 2007, EuCAP 2007, The Second European Conference on 11-16 Nov. 2007, pp. 1-6.

[5] D. Manteuffel, M. Arnold, "Considerations for reconfigurable multi-standard antennas for mobile terminals", Antenna Technology: Small Antennas and Novel Metamaterials, 2008, iWAT 2008. International Workshop in 2008, pp. 231-234

[6] M. R. Hamid, P. S. Hall, P. Gardner, and F. Ghanem, "Frequency reconfigurable Vivaldi antenna", Antennas and Propagation (EuCAP), 2010 Proceedings of the Fourth European Conference in 2010, pp. 1-4

[7] R. Bose, and A. Sinha, "Tunable patch antenna using a liquid crystal substrate", Radar Conference, 2008, RADAR '08, IEEE, 26-30 May 2008, pp. 1-6.

[8] Y. Tawk, J. Costantine, and C. G. Christodoulou, "A rotatable reconfigurable antenna for cognitive radio applications", Radio and Wireless Symposium (RWS), 2011 IEEE, pp. 158-161

[9] V. Plicanic, B. K. Lau, A. Derneryd and Z. Ying, "Actual diversity performance of a multiband diversity antenna with hand and head effect", IEEE Trans. Antennas Propagat., Vol. 57, May, 2009, pp. 1547-1556

[10] H. T. Chattha, Y. Huang, X. Zhu, and Y. Lu, "Dual-feed PIFA diversity antenna for wireless applications", Electronics Letters, Vol. 46, Issue 3, 2010, pp. 189-190

GRADING OF POMEGRANATE LEAF DISEASE

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Abstract- Present paper is an attempt to automatically grade the disease on the Pomegranate plant leaves. This innovative technique would be a boon to many and would have a lot of advantages over the traditional method of grading. There has been a sea change in the mindset and the effort put down by the agricultural industry by adapting to the current trends & technologies. One such example is the use of Information and Communication Technology (ICT) in agriculture which eventually contributes to Precision Agriculture. Presently, plant pathologists follow a tedious technique that mainly relies on naked eye prediction and a disease scoring scale to grade the disease. Manual grading is not only time consuming but also does not give precise results. Hence the current paper proposes an image processing methodology to deal with one of the main issues of plant pathology i.e disease grading by employing Fuzzy Logic. The results are proved to be accurate and satisfactory in contrast to manual grading and hopefully take a strong leap forward in establishing itself in the market as one of the most efficient and effective process.

General Terms*- Leaf diseases, disease grading, image processing, disease management*

Keywords*- Percent Infection, K-means clustering, color image segmentation, Fuzzy logic, machine vision.*

I. INTRODUCTION

Plant diseases can cause significant reduction in crops and lead to poor quality of agricultural products[2]. Research in agriculture is aimed towards increase of productivity and food quality at reduced expenditure and with increased profit [2]. Although there is an industrial recognized corresponding standard to grade the leaf spot disease[5-8], the naked eye observation method is mainly adopted in the production practice. With the advent of new technologies and superior techniques, adopting these means would indeed help this sector to outperform in the coming days. One of the methods is precision agriculture. Precision agriculture aims to optimize field-level management and also provides farmers with a wealth of information to build up a record of their farm; improve decision-making; foster greater traceability; enhance marketing of farm products; enhance the inherent quality of farm products.

Grading of the samples is a daunting task, one of the major reasons being the difference of personal knowledge and practical experience, the same samples are classified into different grades by different experts. Therefore, the result is usually subjective and it is impossible to measure the disease extent precisely as the outcomes may vary and could be misleading.

II. PRESENT GRADING METHOD: MANUAL GRADING

Plants are bound to have diseases. The infected plants are diagnosed and treatment is suggested to cure the disease. To treat the disease chemical pesticides are - used. Pesticides are substances or mixture of substances intended for preventing, destroying, repelling or mitigating any pest. Chemicals are continually becoming a more intricate part of modern society. The rampant use of these chemicals, under the adage, “if little is good, a lot more will be better” has played havoc with human and also on agricultural products. Use of these toxic chemicals can only be minimised when the disease is identified accurately along with the stage in which the disease is; so that only a proper and adequate quantity of pesticide can be used for the treatment rather than blindly spraying it on the infected areas.

Presently,a disease scoring scale is used by the plant pathologists to grade the disease. This is shown in Table 1.

Table1: Disease scoring scale for pomegranate leaves

Percent Infection	Disease Grade
0-5.999	0
6-15.999	1
16-25.999	2
26-35.999	3
36-45.999	4
46-55.999	5
56-65.999	6
66-75.999	7
76-85.999	8
86-100	9

From the table 1, it is observed that the grade of the disease is assigned based on the percent-infection i.e., if the infection percent is about 7 then the grade is 1. Grid paper analysis is presently used to calculate percent-infection.

The main disadvantages of using this method are that it is time consuming and burden of repetitive tasks. To add to the list since human intervention is involved, it is prone to errors which an area of concern.

Due to the ill effects of these methods, present paper suggests a solution to overcome the problems. Machine vision is one such way to grade the disease which can be a boon to the agronomists.

III. METHODOLOGY

The proposed method presents a system that will grade the disease of pomegranate leaves. The flow chart of the process is presented in figure 1. The system is divided into the following steps: (1) Image acquisition (2) Image Pre-processing (3) Color image segmentation (4) Calculating A_T and A_D (5) Disease grading by Fuzzy Logic.

1.1 Image acquisition

The first stage of any vision system is the image acquisition stage. The digitization and storage of an image is referred as the image acquisition. After the image has been obtained, various methods of processing can be applied to the image to perform the many different vision tasks required today.

However, if the image has not been acquired satisfactorily then the intended tasks may not be achievable, even with the aid of some form of image enhancement. All the images are saved in the JPEG format. For the purpose of image acquisition, author has visited and captured images from several pomegranate farms in Rahuri, Ahmed Nagar district, India.

1.2 Image Pre-processing

Preprocessing images commonly involves removing low-frequency background noise, normalizing the intensity of the individual particles images, removing reflections, and masking portions of images[1]. Image preprocessing is the technique of enhancing data images prior to computational processing. Image processing is a form of signal processing for which the input is an image, such as a photograph or video frame; the output of image processing may be either an image or a set of characteristics or parameters related to the image [1].

Pre-processing uses the techniques such as image resize, filtering, segmentation, cropping, etc. Initially, captured images are resized to a fixed resolution so as to utilize the storage capacity or to reduce the computational burden in the later processing. Noise is inevitable during image acquisition or transmission [1]. Noise would disturb the segmentation and the feature extraction of disease spots. So they must be removed or weakened before any further image analysis by applying an appropriate image filtering operation. In the present work, auhor has considered Gaussian filter to filter out the input images.

3.3 Image segmentation

Image segmentation refers to the process of partitioning the digital image into its constituent regions or objects so as to change the representation of the image into something that is more meaningful and easier to analyze.

The level to which the partitioning is carried depends on the problem being solved i.e. segmentation should stop when the objects of interest in an application have been isolated [5].

In the current work, the very purpose of segmentation is to identify regions in the image that are likely to qualify as diseased regions.

There are various techniques for image segmentation. K-means clustering method has been used in the present work to carry out segmentation. K-Means Clustering is a method of cluster analysis which aims to partition n observations into k mutually exclusive clusters in which each observation belongs to the cluster with the nearest mean.

When the segmentation is completed, one of the clusters contains the diseased spots being extracted. This image is saved and considered for calculating AD

3.4 Calculating AT and AD

In image processing terminology area of a binary image is the total number of on pixels in the image[1].

Hence, the original resized image is converted to binary image such that the pixels corresponding to the leaf image are on. From this image total leaf area (A_T) is calculated. Similarly, the output image from color image segmentation, containing the disease spots, is used to calculate total disease area (A_D)[1].

3.5 Disease grading by Fuzzy Logic

Once A_T and A_D are known, the percent-infection (PI) is calculated by applying the formula (1).

$$PI= (A_D / A_T) *100 \quad . . . (1)$$

Now that, the Grade of the disease has to be determined from PI. , Fuzzy Logic has been employed for this purpose.

FLOWCHAR

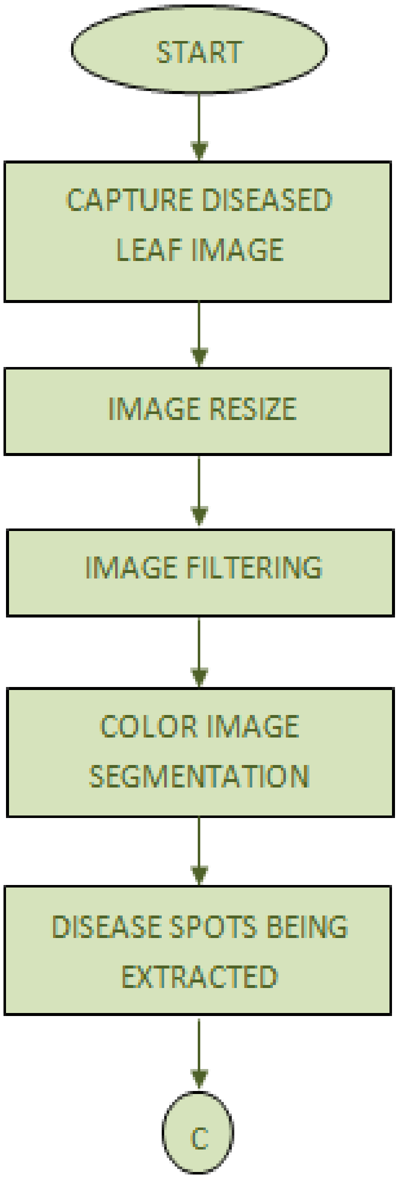


Figure1: Flowchart of the process

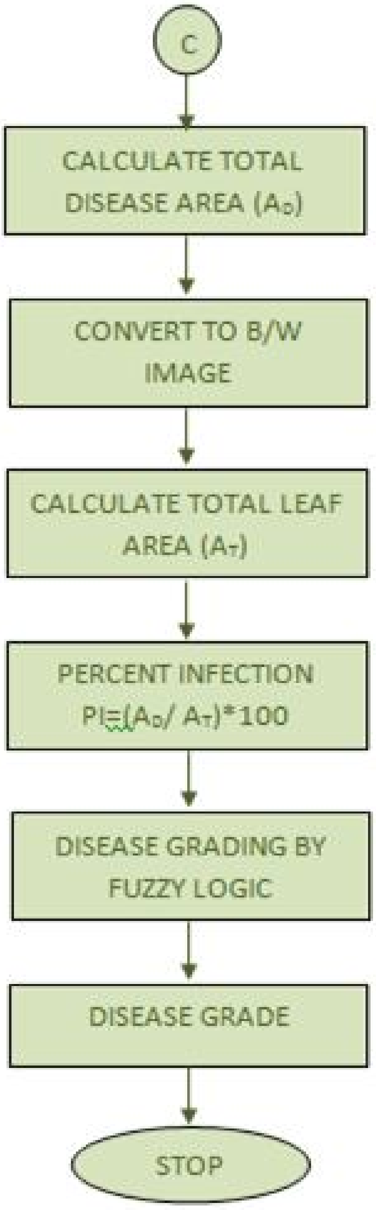


Figure 1: FlowChart of the complete Process[1]

IV. RESULTS AND ANALYSIS

4.1 Image acquisition

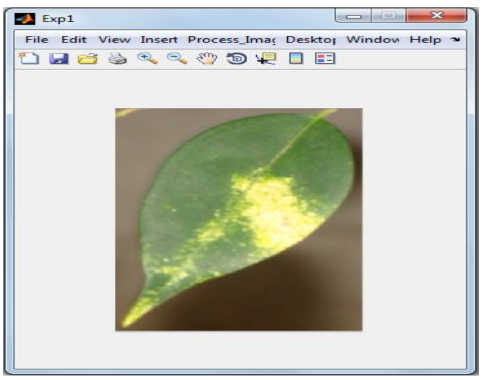


Figure 2 shows the query image.

4.2 Image Pre-processing

4.2.1 Resize The image is resized to a resolution of [200 250].

4.2.2 Filtering Gaussian filters are used to remove noise

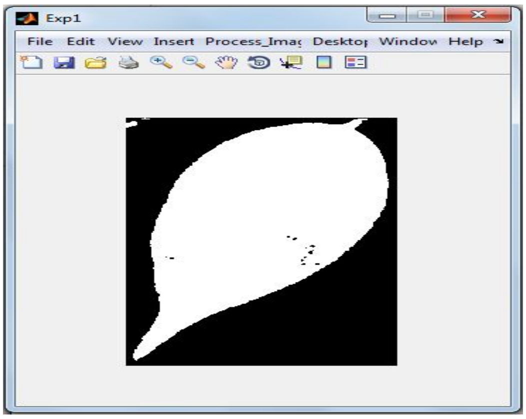
4.3 Color image segmentation

K-means segmentation algorithm requires users to select the value 'k'. The correct choice of k is often ambiguous. Increasing k will always reduce the amount of error in the resulting clustering, to the extreme case of zero error if each data point is considered its own cluster (i.e., when k equals the number of data points, n). Intuitively then, the optimal choice of k will strike a balance between maximum compression of the data using a single cluster, and maximum accuracy by assigning each data point to its own cluster. After some trial and error method, for the current work, value of K is chosen as 10

4.4 Calculating A_T and A_D

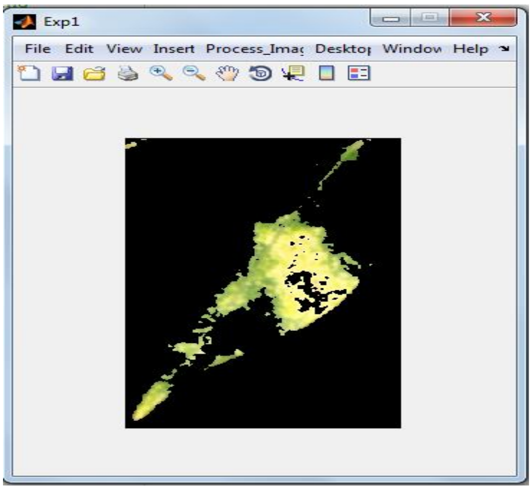
Total Leaf Area (AT)

Figure 3 shows the binary image of the original resized image.



Total Disease Area (AD)

Figure 4 shows the binary image of the disease portion.



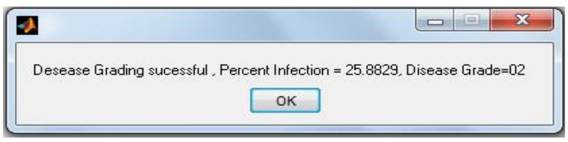
4.5 Disease grading by Fuzzy Logic

From (1), Percent-infection is given by

PI= (AD / AT) *100

= (34584 / 133617) * 100

= 25.8829 %



A Fuzzy Inference System (FIS) is developed for disease grading by referring to the disease scoring scale in Table 1. For the FIS for disease grading, input variable is Percent Infection and output variable is Grade. Triangular membership functions are used to define the variables and eleven fuzzy rules are set to grade the disease. Figure4 shows the result of grading. From the result, it can be observed that the accurate values of percent-infection and disease grade are obtained with which a proper treatment advisory can be given thereby eliminating the above mentioned problems.

V. CONCLUSION

The main motive of this paper is to improve the efficiency and productivity through a robust system which can overcome the shortcomings of the manual process. Looking at the current scenario an approach to automatically grade the disease on plant leaves is very much essential. As discussed in the earlier sections Grading System built by Machine Vision and Fuzzy Logic is very useful for grading the disease. The disadvantages faced through the manual grading would be overcome once this system is adopted and will help the pathologists in terms of complexity and time. Through research and experiments it has been observed that the results found are precise, accurate and acceptable.

REFERENCES

[1] Sanjeev S Sannakki1, Vijay S Rajpurohit, V B Nargund, ArunKumar R, Prema S Yallur , “Leaf Disease Grading by MachineVision and Fuzzy Logic”, Int. J. Comp. Tech. Appl., Vol 2 (5),1709-1716, 2011.

[2] Santanu Phadik ar and Jaya Sil (2008) Proceedings of 11th International Conference on Computer and Information Technology (ICCIT 2008)

[3] Camargo, A. and Smith, J. S. (2009). An image-processing based algorithm to automatically identify plant disease visual symptoms,Biosystems Engineering, 102(1) pp. 9-21

[4] Narendra V.G., Hareesh K.S. (2010). Prospects of Computer Vision Automated Grading and Sorting Systems in Agricultural and Food Products for Quality Evaluation , International Journal of Computer Applications, 2, 1.

[5] Nur Badariah Ahmad Mustafa, Syed Khaleel Ahmed, Zaipatimah Ali, Wong Bing Yit, Aidil Azwin Zainul Abidin, Zainul Abidin Md Sharrif, “Agricultural Produce Sorting and Grading using Support Vector Machines and Fuzzy Logic”, 2009 IEEE International Conference on Signal and Image Processing Applications

[6] Rafael C.Gonzalez & Richard E.Woods, “Digital Image Processing”, Second edition, 2005.

[7] Üsmail Kavdir,Daniel E. Guyer, “Apple Grading Using Fuzzy Logic”, Turk J Agric (2003), 375-382 © T.BÜTAK

[8] Meunkaewjinda, Kumsawat, At-takitmongcol, k.; and Srikaew, A. “Grape leaf disease detection from color imagery using hybrid intelligent system”. Electrical Engineering/ Electronics, Computer, Telecommunications and Information Technology, 2008. ECTI-CON 2008. 5th International Conference on v.1, pp.513–516, 14-17 May, 2008.

MICROMINIATURE SPY-BOT USING ZIGBEE

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Abstract- With evolving technology in the 21st century, the need for more machine controlled devices to reduce human involvement in risky operations becomes necessary. As robot comes closer to humans, an efficient human-robot-control is utmost necessary. In this technical paper, we intend to provide accurate information regarding spybot, which is controlled by Zigbee andPIC microcontroller, through our research.

Keywords- *Zigbee,Spybot*

I. INTRODUCTION

We propose a new type of Micro-miniature sized Spy Robot which can be controlled from anywhere around the network area. Just by using a ZIGBEE enabled device, the user can control the Spy Robot for its various utilities.The main task of this project is to attach the XBee module3 to the JOP powered LEGO robot4 and implement a simple example protocol. The purchased XBee Starter Kit5 contains two XBee Modules, one serial interface board, one USB interface board, one installation CD, and other necessary accessories such as cables, adapters, etc.

The CD includes X-CTU software for windows which allows convenient configuration of the XBee Modules. Without the software, the Modules also can be configured through a terminal program in AT Command Mode. As shown in Figure 16, the base XBee module can be connected sprightly e.g. to a PC through the USB interface, whereas the remote XBee Module shall be directly wired to the LEGO robot which requires some soldering and hardware adaption in VHDL[IV]. With the hardware placement, the interpretation of remote control signals is realized in software.

II. OVER ALL BLOCK DIAGRAM

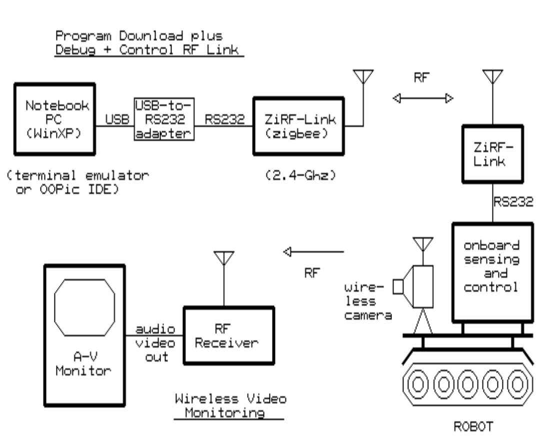

Figure 1: Block Diagram of overall system[VI].

Command Interface - we wrote a simple command interface for the Pic which allows us to send simple movement commands via the Zigbee downlink, and receive back status info via the Zigbee uplink. On a terminal emulator, or using the Pic IDE common window, we can press single keyboard keys to command the robot. These include:

"F"=forward, "B"=back-up, "A"=accelerate, "D"=decelerate, "L"=turn-left, "R"=turn-right, and "S"=stop. We also added the ability to remotely move the video camera in pan tilt dimensions on the servo mount, in order to better follow the robot's behaviour. Commands are sent over the RF, and then the robot sends back status info, such as left-right motor speed and direction, as well as battery voltage plus various sensor readings. We can watch the robot directly, or monitor its behaviour remotely via the wireless video uplink. For now, we won't worry about doing a more complex software console for the PC as, first, that would be a major undertaking, and secondly, we only want rudimentary remote control of the spy robot, as the main purpose is to develop software for autonomous operation of the robot. Rather than control the robot ourselves, it will be much more fun to watch the robot, and reload its internal parameters, while it makes most of its own decisions.

A. BLOCK DIAGRAM OF TRANSMITTER [VI]

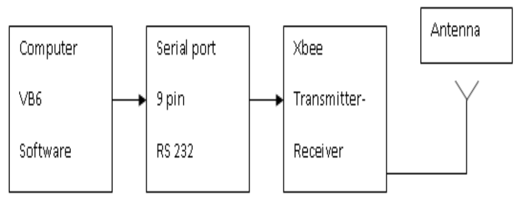

Figure 2: Block Diagram of transmitter.

By using the software visual basic 6(VB6) we transmit the signal to the serial port, this serial port is of 9 pin which is RS 232 and finally transmitted to ZigBee transreciever.

ZigBee Transmitter Receiver Module

Figure 3: ZIGBEE chip

B.BLOCK DIAGRAM ON ROBOT SIDE[VI]

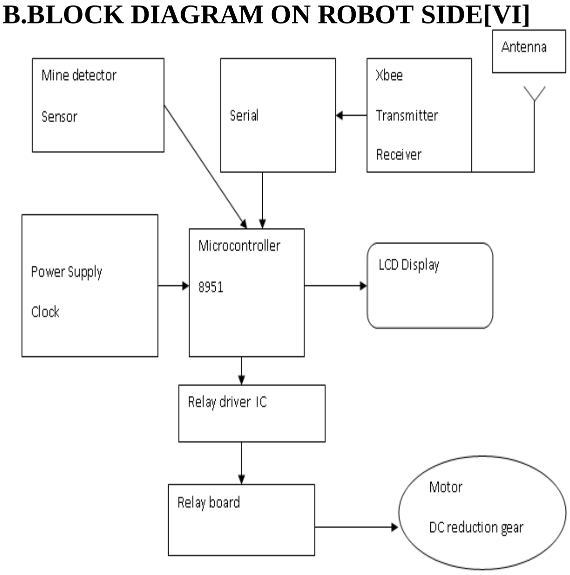

Figure 4: Block Diagram of ROBOT side

a. ZIGBEE PROTOCOL:

The ZigBee protocol was engineered by the ZigBee Alliance, a non-profit consortium of leading semiconductor manufacturers, technology providers, OEMs and end-users worldwide. The protocol was designed to provide OEMs and integrators with an easy-to-use wireless data solution characterized by low-power consumption, support for multiple network structures and secure connections[III].

b. ZIGBEE ADVANTAGE:

The ZigBee protocol was designed to carry data through the hostile RF environments that routinely exist in commercial and industrial applications[IV].

c. ZIGBEE PROTOCOL FEATURES:

- Low duty cycle - Provides long battery life
- Low latency
- Support for multiple network topologies: Static, dynamic, star and mesh
- Direct Sequence Spread Spectrum (DSSS)
- Up to 65,000 nodes on a network
- 128-bit AES encryption – Provides secure connections between devices
- Collision avoidance
- Link quality indication
- Clear channel assessment
- Retries and acknowledgements

d. PIC MICROCONTROLLER

In order to enable the microcontroller to operate properly it is necessary to provide:

- Power supply;
- Reset signal; and
- Clock signal.

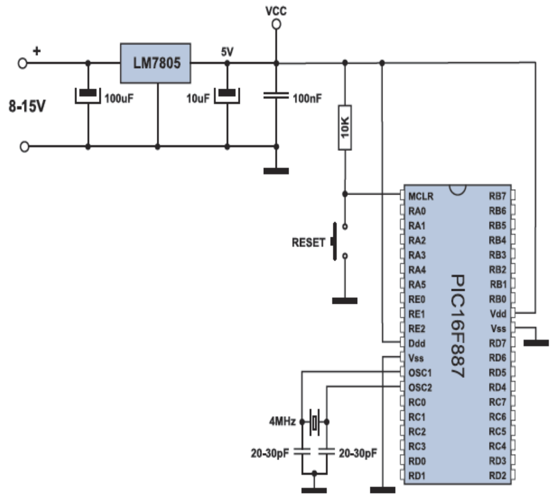

Figure 5:PIC16F887 pin configuration

POWER SUPPLY

Even though the PIC16F887 can operate at different power supply voltages. A 5V DC power supply voltage is the most suitable[V]. The circuit, shown on the top, uses a cheap integrated three-terminal positive regulator LM7805 and provides high quality voltage stability and quite enough current to enable the microcontroller and peripheral modules to operate normally (enough means 1A).

RESET SIGNAL

A push button connecting the MCLR reset pin to GND is not necessary, but is almost always provided as it enables the microcontroller to recover fast if something goes wrong. By pressing this button, the MCLR pin is supplied with 0V, the microcontroller reset occurs and the program execution starts from the beginning[V]. A 10K resistor is used to prevent shortening the 5V DC rail to earth from occurring when the RESET button is pressed.

CLOCK SIGNAL

The oscillator can be run in four different modes[V]:

- LP - Low Power Crystal;
- XT - Crystal / Resonator;
- HS - High speed Crystal / Resonator; and
- RC - Resistor / Capacitor

III. SPYROBOT

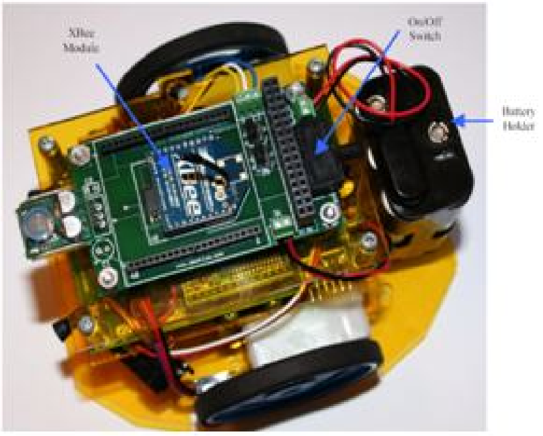

Figure 6: Top view of SPY-BOT

IV. APPLICATION

During war or conflict, human spies are required to get the necessary information in order for a nation to gain an upper hand over the other (espionage)[I]. However, this task jeopardizes human lives (spies). In order to prevent this, micro sized Robot can be used to extract vital information without any casualties[I]. Apart from that, this robot can also be utilized for rescue operations during any natural disaster, such as earthquake, where humans stuck under the debris can be determined.

V. CONCLUSION

This pocket sized Spy robot consists of a micro camera mounted on it. The controller can move the robot using ZIGBEE wireless network. The image from the camera on the robot is visible to the user from the LCD screen on the wireless controller. The image obtained can be used to obtain vital information used for Espionage or rescue operation. Minimum distance obtained using ZIGBEE wireless controller is 100mtr.

REFERENCES

[1]. I.T.Jolliffe,"Principle Component Analysis",Newyork: Spring.Verlag.2002.

[2]. L.Jung, "System identification: theory for user", uppersaddle river, NJ: Prentice-hall, 1999.

[3]. Theodore S. Rappaport,"Wireless communication", uppersaddle river,NJ: Prentice-hall, 2002.

[4]. Shahin Farhani,"Zigbee wireless network and trans-receiver", San diego california, 2008.

[5]. John Iovine,"PIC Microcontroller project book", MC-graw hill/tab electronics, 2004.

[6]. Dan Marzollo,"I spy runaway robot", Milan, 2003

[7]. James Holloman,"surface mount technology for pc boards", prompt publication, 1995.

[8]. Mike Judd,"Soldering in electronics assembly", Butterworth heinemann, 1999.

AUTONOMOUS INTELLIGENT ROBOT

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Abstract- Robotics is part of Today's communication. ROBOT has sufficient intelligence to cover the maximum area of provided space. It has a infrared sensor which are used to sense the obstacles coming in between the path of ROBOT. It will move in a particular direction and avoid the obstacle which is coming in its path

Keywords- *Zigbee, Autonomous, Intelligent.*

I. INTRODUCTION

Autonomous Intelligent Robots are robots that can perform desired tasks in unstructured environments without continuous human guidance. The robot can be accessed wirelessly i.e. technician can directly order the robot. To avoid collisions, a proximity detector has been added which causes robot to stop as soon as it encounters an obstacle in its way, thus avoiding accidents.

A fully autonomous robot has the ability to

- Gain information about the environment.
- Work for an extended period without human intervention.
- Move either all or part of itself throughout its operating environment without human assistance.
- Avoid situations that are harmful to people, property, or itself unless those are part of its design specifications.

Autonomous robot for path finding and obstacle evasion is a vehicle, which is used to follow the path autonomously. This robot can be operated in manual mode, in case human guidance is necessary.

II. OVERALL BLOCK DIAGRAM

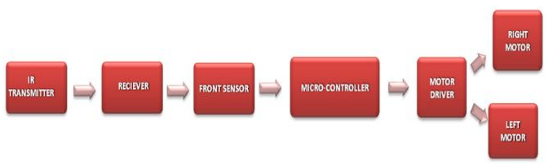

Figure 1: Block Diagram of overall system.

The information is processed by the micro-controller and is given to the motor driver chip and this enables the motor operations accordingly.

Commands are sent over the RF, and then the robot sends back status info, such as left-right motor speed and direction, as well as battery voltage plus various sensor readings. Our project is for human-less interface. When the system is turned ON, it will enable the AT89C51 Micro-controller to turn on the Motor driver and the camera system. The Autonomous system is working with an array of sensors and switching circuits. This array of sensors will be an IR transmitter and receivers. This information is processed by the micro-controller and is given to the motor driver chip and this enables the motor operations accordingly.

A. BLOCK DIAGRAM OF TRANSMITTER AND RECIEVER

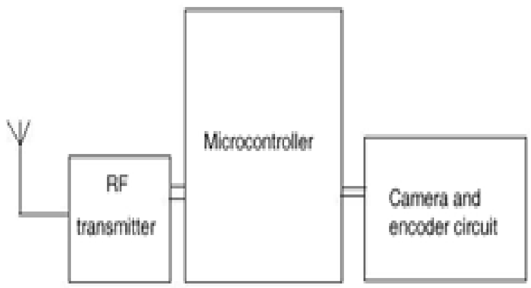

Figure 2: Block Diagram of transmitter.

We have wireless video transmitter receiver circuits that have a mini camera installed on our autonomous vehicle and this camera is connected with a RF transmitter via serial and micro-controller. It is received by RF receiver and is decoded and sent to the computer through serial port RS232. We are transmitting analog video signal through digital RF trans-receiver.

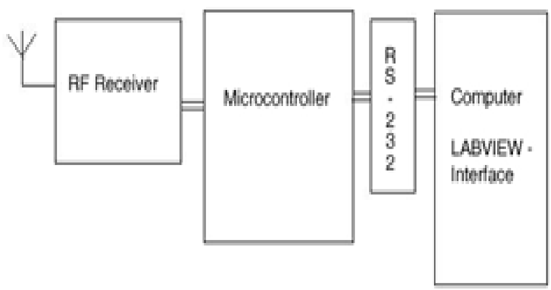

Figure 3: Block Diagram of receiver.

ZigBee Transmitter Receiver Module

Figure 4: ZIGBEE chip

Command Interface - we developed a simple set of commands for the microcontroller which allows us to send simple instructions via the Zigbee downlink, and receive back status info via the Zigbee uplink. On a terminal emulator, or using the microcontroller IDE common window, we can press single keyboard keys to command the robot.

a. ZIGBEE PROTOCOL:

The ZigBee protocol was engineered by the ZigBee Alliance, a non-profit consortium of leading semiconductor manufacturers, technology providers, OEMs and end-users worldwide. The protocol was designed to provide OEMs and integrators with an easy-to-use wireless data solution characterized by low-power consumption, support for multiple network structures and secure connections [III].

b. ZIGBEE ADVANTAGE:

The ZigBee protocol was designed to carry data through the hostile RF environments that routinely exist in commercial and industrial applications [IV].

c. ZIGBEE PROTOCOL FEATURES:

- Low duty cycle - Provides long battery life
- Low latency
- Support for multiple network topologies: Static, dynamic, star and mesh
- Direct Sequence Spread Spectrum (DSSS)
- Up to 65,000 nodes on a network
- 128-bit AES encryption – Provides secure connections between devices
- Collision avoidance
- Link quality indication
- Clear channel assessment
- Retries and acknowledgements

d. PIC MICROCONTROLLER

The AT89C51 is a low-power, high-performance CMOS 8-bit microcomputer with 4K bytes of Flash programmable and erasable read only memory (PEROM). The device is manufactured using Atmel's high-density nonvolatile memory technology and is compatible with the industry-standard MCS-51 instruction set and pinout. The on-chip Flash allows the program memory to be reprogrammed in-system or by a conventional nonvolatile memory programmer.

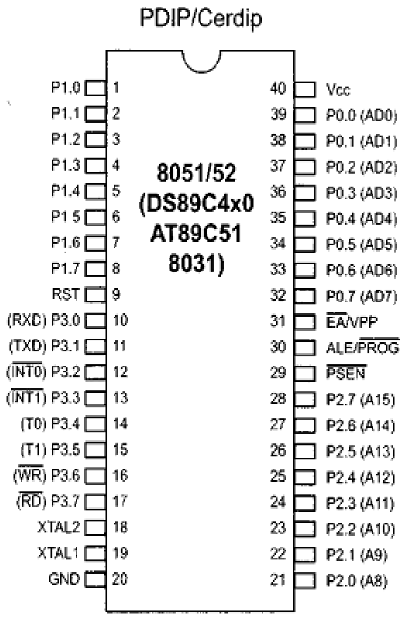

Figure 5:AT89C51 pin configuration

RST

Reset input. A high on this pin for two machine cycles while the oscillator is running resets the device.

ALE/PROG

Address Latch Enable output pulse for latching the low byte of the address during accesses to external memory. This pin is also the program pulse input (PROG) during Flash programming.

Idle Mode

In idle mode, the CPU puts itself to sleep while all the on chip peripherals remain active. The mode is invoked by software. The content of the on-chip RAM and all the special functions registers remain unchanged during this

Mode. The idle mode can be terminated by any enabled

Interrupt or by hardware reset.

Features

- Compatible with MCS-51™ Products
- 4K Bytes of In-System Reprogrammable Flash Memory
- Endurance: 1,000 Write/Erase Cycles
- Fully Static Operation: 0 Hz to 24 MHz
- Three-level Program Memory Lock
- 128 x 8-bit Internal RAM
- 32 Programmable I/O Lines
- Two 16-bit Timer/Counters

- Six Interrupt Sources
- Programmable Serial Channel
- Low-power Idle and Power-down Modes

III. AUTONOMOUS ROBOT

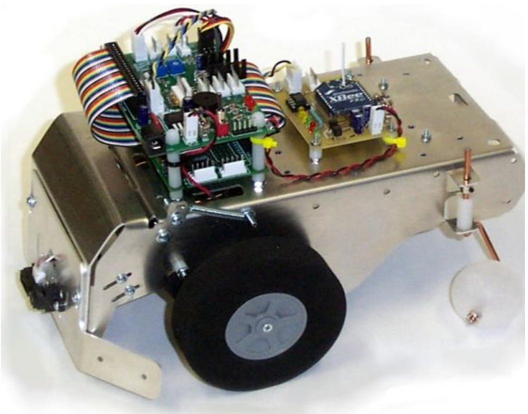

Figure 6: AUTONOMOUS ROBOT

IV. APPLICATION

For computing safe and unsafe areas on the surface within that field of vision. Driving the Rover along the planned path. Maintenance of nuclear plants. Military applications, such as reconnaissance flying drones. Pick up and move activities Pick up and move activities.

On top of obstacle sensors temperature/pressure sensors can be added to monitor the atmospheric conditions around. This is useful in places where the environment is not suitable for humans.

V. CONCLUSION

This pocket sized Spy robot consists of a micro camera mounted on it. The controller can move the robot using ZIGBEE wireless network. The image from the camera on the robot is visible to the user from the LCD screen on the wireless controller. The image obtained can be used to obtain vital information used for Espionage or rescue operation. Minimum distance obtained using ZIGBEE wireless controller is 100mtr.

REFERENCES

1. I.T.Jolliffe,"Principle Component Analysis",Newyork: Spring.Verlag.2002.
2. L.Jung, "System identification: theory for user", uppersaddle river, NJ: Prentice-hall, 1999.
3. Theodore S. Rappaport,"Wireless communication", uppersaddle river,NJ: Prentice-hall, 2002.
4. Shahin Farhani,"Zigbee wireless network and trans-receiver", San diego california, 2008.
5. John Iovine,"PIC Microcontroller project book", MC-graw hill/tab electronics, 2004.
6. Dan Marzollo,"I spy runaway robot", Milan, 2003
7. James Holloman,"surface mount technology for pc boards", prompt publication, 1995.
8. Mike Judd,"Soldering in electronics assembly", Butterworth heineman, 1999

CHANNEL ESTIMATION ALGORITHMS OF OFDM IN VARIOUS SYSTEMS: A REVIEW

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Abstract- Orthogonal frequency-division multiplexing (OFDM) effectively mitigates inter symbol interference (ISI) caused by the delay spread of wireless channels. Therefore, it has been used in many wireless systems and adopted by various standards. In this paper, we present a comprehensive survey on channel estimation in OFDM for wireless communications as channel estimation is a very important issue for wireless communication. In this paper it is given an overview of the basic principle of OFDM and review of different algorithm proposed in last few years.

Index terms*- orthogonal frequency division multiplexing (OFDM), channel impulse response (CIR), Cyclic prefix(CP), interchannel interference(ICI),Raised cosine(RC)*

I.INTRODUCTION

Orthogonal frequency division multiplexing (OFDM) is a transmission technique that is built-up by many orthogonal carriers that transmits simultaneously. The main idea behind OFDM is that a signal with a long symbol duration time is less sensitive to multipath fading, than a signal with a short symbol time. Hence, a gain in performance can be achieved by sending several parallel symbols with a long symbol time than sending them in a series with a shorter symbol time. The basic technique of OFDM has been known for around 40 years, but it is not since recently that OFDM has become widely used. Products that use OFDM are for example WiMAX, WLAN (Wireless Local Area Network) 802.11, x-DSL (x- Digital Subscriber Line) and DVT (Digital Video Broadcasting).The basic principle of OFDM is to split a high rate data-stream into multiple lower rate data streams that are transmitted simultaneously over a number of sub carriers. OFDM sends multiple high-speed signals concurrently on orthogonal carrier frequencies. This results much more efficient use of bandwidth as well as robust communications during noise and other interferences. With OFDM, it is possible to have overlapping sub channels in the frequency domain, thus increasing the transmission rate. In order to avoid a large number of modulators and filters at the transmitter and complementary filters and demodulators at the receiver, it is desirable to be able to use modern digital signal processing techniques, such as fast Fourier transform (FFT).

The major advantages of OFDM are its ability to convert a frequency selective fading channel into several nearly flat fading channels and high spectral efficiency. The rest the paper is organized as follows: In section II, literature review has been described. In section III, principle of OFDM system has been described. In section IV, paper review has been described. In section V, conclusion has been described.

II.LITERATURE REVIEW

The origins of OFDM development started in the late 1950’s with the introduction of Frequency Division Multiplexing (FDM) for data communications. In 1966 [1] Chang patented the structure of OFDM and published the concept of using orthogonal overlapping multi-tone signals for data communications. In 1971 Weinstein and Ebert [2] introduced the idea of using a Discrete Fourier Transform (DFT) for implementation of the generation and reception of OFDM signals, eliminating the requirement for banks of analog subcarrier oscillators. This presented an opportunity for an easy implementation of OFDM, especially with the use of Fast Fourier Transforms (FFT), which are an efficient implementation of the DFT. This suggested that the easiest implementation of OFDM is with the use of Digital Signal Processing (DSP), which can implement FFT algorithms [3]. It is only recently that the advances in integrated circuit technology have made the implementation of OFDM cost effective.

There are many algorithms and models are studied and proposed for channel estimation in more efficient and optimum way. In this paper we have consider [6] to [14] for channel estimation review.

III. THE PRINCIPLE OF OFDM

Orthogonal Frequency Division Multiplexing (OFDM) is a multicarrier transmission technique, which divides the bandwidth into many carriers; each one is modulated by a low rate data stream. In term of multiple access technique, OFDM is similar to FDMA in that the multiple user access is achieved by subdividing the available bandwidth into multiple channels that are then allocated to users [4]-[5] However, OFDM uses the spectrum much more efficiently by spacing the channels much closer together. This is achieved by making all the carriers orthogonal to one another, preventing interference between the closely spaced carriers. The figure shows the difference between the conventional non-overlapping multicarrier technique and overlapping multicarrier modulation technique. As shown in figure 2, by using the overlapping multicarrier modulation technique, we save almost 50% of bandwidth. To realize the overlapping multicarrier technique, however we need to reduce crosstalk between subcarriers, which means that we want orthogonality between the different modulated carriers.

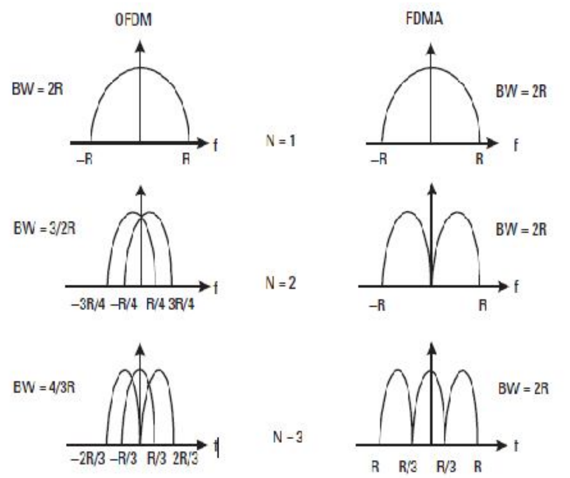


Fig 1:

Concept of OFDM Signal: Orthogonal Multicarrier Technique Versus Conventional Multicarrier Technique The orthogonality of the carriers means that each carrier has an integer number of cycles over a symbol period. Due to this, the spectrum of each carrier has a null at the center frequency of each of the other carriers in the system. This results in no interference between the carriers, allowing then to be spaced as close as theoretically possible. This overcomes the problem of overhead carrier spacing required in FDMA. Each carrier in an OFDM signal has a very narrow bandwidth (i.e.1kHz), thus the resulting symbol rate is low. This results in the signal having a high tolerance to multipath delay spread, as the delay spread must be very long to cause significant inter-symbol interference (e.g. > 500 μsec).

As previously mentioned an OFDM system has the ability to completely remove ISI between two OFDM symbols. A simple solution, which would remove the ISI, is to simply insert a guard interval between the OFDM symbols, i.e. simply wait for all the multipaths reflections of the transmitted OFDM symbol to fade out before transmitting another OFDM symbol, as in Figure 2.

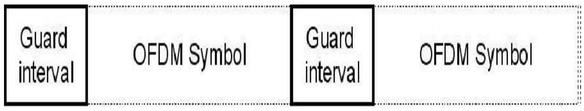


Fig2: Remove ISI through guard interval

\There is however a problem using guard intervals to remove the ISI, which has do with the property of the DFT. The DFT is cyclic, so if the received OFDM symbol is not cyclic, it will cause ICI (Inter Carrier Interference) between the subcarriers. The solution is to add a cyclic extension to the OFDM symbol before transmitting it. The cyclic extension that is added before the transmission is simply the end part of the OFDM symbol that has been copied and transmitted before the OFDM symbol as in Figure 3. The cyclic extension is referred to as CP (Cyclic Prefix).

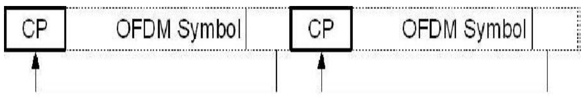


Fig 3: Cyclic Prefix

The length of the CP is set to at least to the maximum length of multipath delay of the radio channel, i.e. to at least the same number of taps as the channel.

IV. PAPER REVIEW

A. Songping wu and Yeheskel bar-ness “OFDM channel estimation in the presence of frequency offset and phase noise”IEEE 2003.

CP based frequency estimation in[7] is quite simple and bandwidth efficient,but it doesnot consider the effects of multipath fading and estimation results may not be accurate as it is based on CP of one symbol only.method proposed in [8] improves [7]by considering the CIR length and taking more symbols into account. However ,when averaging frequency offset estimates obtained separately from each symbol,accumulated errors may be larger than expected.

A different method which solves these problem was proposed by Songping Wu and Yeheskel Bar-Ness with aid of CP based frequency offset estimation statistics independent channel estimation based on [8],several symbols are used to estimate frequency offset ,but it does not accumulate errors by using the following expression

$$\hat{\varepsilon}(p)=\frac{1}{2\pi}angle\left(\frac{1}{N_g\cdot M}\sum_{m=0}^{M-1}\sum_{k=-N_g+p}^{-1}\left(r_m^*(k)r_m(k+N)\right)\right) \qquad (1)$$

Where P is CIR length and unknown so it can be initially set to 1 and found by iteration.

The minimization of channel estimation errors leads to the most likely CIR length ,which is then used to optimize frequency offset estimate and also used to filter the LS channel estimate reducing its sensitivity to noise and ICI. The most likely CIR length can be found by minimizing the cost function

$$\left\| \mathbf{h} - \mathbf{h}_p^{ls} \right\|^2 \quad (2)$$

Where h is channel impulse response and $\mathbf{h}_p^{ls}$ is channel estimate using LS method. Minimization of(2) can be obtained by the first occurrence of the minimum of

$$\left\| \mathbf{h}_p^{ls} - \mathbf{h}_{p-1}^{ls} \right\|^2 \quad (3)$$

Minimum of (3) occurs with high probability at the point where p is equal to the CIR length. And solution presented the most likely CIR length can be found by varying p between 1 and N_g ,and choosing the values which satisfies the following criteria

$$\left\| \mathbf{h}_p^{ls} - \mathbf{h}_{p-1}^{ls} \right\|^2 \le \left\| \mathbf{h}_{p-1}^{ls} - \mathbf{h}_{p-2}^{ls} \right\|^2$$

and

$$\left\| \mathbf{h}_p^{ls} - \mathbf{h}_{p-1}^{ls} \right\|^2 \le \left\| \mathbf{h}_{p+1}^{ls} - \mathbf{h}_{p}^{ls} \right\|^2 \quad (4)$$

This paper also contained the solution in case of interpolated Pilot symbols where pilot signal are multiplexed into transmitted data streams. Simulation results contained performance of frequency offset estimation, estimated CIR length, for non interpolation and interpolation case excellent performance has achieved.

B. Xianbin Wang, Yiyan Wu and Jean-Yves Chouinard “Modified Channel Estimation Algorithms for OFDM Systems with Reduced Complexity” IEEE 2004.

The importance of channel estimation is of crucial importance, especially for time-varying channel, where it has to be performed periodically or even continuously. In order to simplify the OFDM channel estimator , a channel length estimation based on second order statistics of the received signal is proposed. Many algorithm using ML & MMSE technique is proposed for channel estimation but system complexity & computation complexity & computation complexity still remains. Such as in [10] a low rank approximation to the frequency domain linear MMSE estimator was proposed , making use of singular value decomposition (SVD) technique which itself is very complicated and it also requires knowledge of the channel frequency correlation and the operating SNR. So in this paper simplification of the ML & MMSE channel estimators for OFDM has been investigated based on second order statistics of the received signal and channel length is estimated as follows.

The auto correlation function R_{yy} of the received signal y has to be determined

$$R_{yy}(m) = E\left\{ y(n) y^*(n+m) \right\}$$

$$= \sum_{k=-\infty}^{\infty} h(k) \sum_{l=-\infty}^{\infty} h^*(l) \cdot E\left\{ x(n-k) x^*(n+m-l) \right\} \quad (5)$$

$$= \sum_{k=-\infty}^{\infty} h(k) R_1(m+k)$$

where

$$R_1(m+k) = \sum_{l=-\infty}^{\infty} h^*(l) \cdot R_{xx}(m+k-l) = R_{xx}(m+k) \otimes h^*(m+k)$$

Denote m+k=r

$$R_{yy}(m) = \sum_{k=-\infty}^{\infty} h(k) R_1(m+k) = \sum_{k=-\infty}^{\infty} h(r-m) R_1(r) = R_1(m) \otimes h(-m)$$

The above equation simplifies as

$$R_{yy}(m) = R_1(m) \otimes h(-m) = R_{xx}(m) \otimes R_{hh^*}(m) \quad (6)$$

Where $R_{hh^*}(m) = h(m)^* \otimes h(-m)$

denotes the correlation function of channel impulse response & its time domain inverted version. Auto correlation function of OFDM signal can be approximated by the kroneckar delta function & equation (6) become

$$R_{yy}(m) \approx R_{hh^*}(m) \otimes \delta(m) = R_{hh^*}(m) \quad (7)$$

Therefore the duration of channel impulse response can be very simply estimated using the R_{yy}.

Algorithm proposed:

1)To determine the channel length from the correlation function (from equation 6). A time selection window with a predetermined maximum channel length was multiplied to the correlation function from (6).

2)The window size was then gradually reduced sample by sample. If the amplitude of sample is less than predetermined threshold then sample will be discarded. This process will continue until the first significant peak in the auto correlation function is found.

3)The channel length is the duration from the central peak to this significant non–zero peak in the correlation function.

4)One way to reduce the amount of computation is to preset a longest expected channel length L* & calculate the auto correlation over this limited range. The amount of complex multiplication needed for the channel length estimation is NxL*.

The proposed schemes in this paper is concluded such that ,amount of computation for channel estimation is reduced and very robust to frequency offset as well as no preamble training sequence or preliminary channel estimation is needed.

C. Xuing Yu,ping Lin, Zhiqiang He and Weilig Wu “OFDM channel estimation with Impulse noise cancellation” IEEE 2007.

Proposed a different approach rather than using the robust estimator [12],estimates a sufficient statistics of the impulse noise and cancel its impact to the time domain least square channel estimate to produce a better channel estimate and can work under a wide range of signal to noise ratio and summarize their proposed time domain OFDM channel estimation with impulse noise cancellation algorithm as follows. Assumptions : CP length(N_p) is larger than the channel length(L),guassian mixture model for additive noise.

1) Get least square time domain channel estimate;

$$\hat{\mathbf{h}} = (\mathbf{X}\mathbf{W}_2)^{\dagger}\,\mathbf{Y}_P .$$

X-N_p N_p matrix with its diagonal elements being the pilot values x_n.
W-$W_{(1:N,1:L)}$ (NXN DFT matrix)
W_1-$W_{(1:N,1:L)}$
W_2-$W_{(P1:N1:PNp;1:L)}$
W_3-$W_{(P1:N1:PNp;1:L)}$
P_1:N_1:P_{Np}-suppose there are N_p pilots located at locations $\{P_n:1\le n\le N_p\}$ with known values $\{x_n;\ 1\le n\le N_p\}$ and equal interval of N_1.
Y_p:frequency response at N_p pilot subcarriers.
h:time domain channel impulse response .
:estimated time domain channel impulse response.
2) Calculate difference;

$$r=Y_p-XW_2$$

3)Estimate V^*;

$$^*=\text{IFFT (r)}$$

4) Prune * to get $\hat{\hat{\mathbf{v}}}^*_i$ according to

$$\hat{\hat{\mathbf{v}}}^*_i = \begin{cases} \hat{\mathbf{v}}^* & \left|\hat{\mathbf{v}}^*\right| > threshold \\ 0 & \left|\hat{\mathbf{v}}^*\right| \le threshold. \end{cases}$$

5) Cancel the impact of estimated impulse noise component on to get new channel estimate.

$$\hat{\hat{\mathbf{h}}} = \hat{\mathbf{h}} - (\mathbf{X}\mathbf{W}_2)^{\dagger}\,\mathbf{W}_4\hat{\hat{\mathbf{v}}}^* .$$

Simulation results provide improved performance in comparison to MSE performance for different values of noise statistics and concluded with very low computation overhead and uses IFFT and threshold blanking to cancel the impact of noise.

D. Guanghui Liu, Liaoyuan Zeng, Hongliang Li, and Zhengning Wang "Complex Coefficient Interpolation Based Channel Estimation for OFDM in Single – Frequency Networks" IEEE 2012

A complex lowpass interpolation based channel estimation scheme is presented for the pilot-aided OFDM system. Classical methods of filter design are used to develop the complex interpolator aiming for countering long-delay echoes in SFN channels. The interpolation is performed as

$$\hat{H}[k] = \sum_{n=-Q}^{Q} b[n]\,H_{up}[k-n]$$

where b[n] is the coefficient of the interpolator with length (2Q + 1).The FDI design is essentially the optimization of the coefficient b[n] according to the characteristics of up[k], consisting of low-rate CFR estimates at pilots separated by (D -1) zeros. Analytically, the IFFT of up[k] produces a periodic replication of the original CIR at a rate of Tu / D, as illustrated in Fig. 4, where the original CIR samples range from 0 to τ_{max}. The job of the FDI is to remove all of the CIR replicas, except those that are multiples of Tu. As we know, if pilot symbols are inserted in the frequency direction at a rate higher than Nyquist rate of the channel, i.e. D < Tu / τ_{max}, an ideal FDI can be derived theoretically to perfectly recover the CFR at data cells, neglecting the effect of noise. In other words, for the OFDM system with the pilot spacing D, the FDI is robust in the multipath channel with

$$\tau_{\max} < \frac{T_u}{2D} \qquad (8)$$

Otherwise, the FDI cannot separate the original CIR from its replicas, and the CE accuracy is influenced by the aliasing effect, degrading the performance of OFDM receiver. In SFN, the transmitters simultaneously send the same content over the same frequency, artificially creating long delay echoes. Thus, the conventional real-coefficient FDI may not meet the OFDM system's requirement in SFN, as indicated by (8).

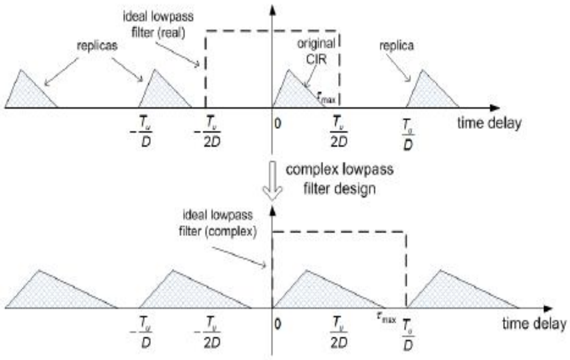
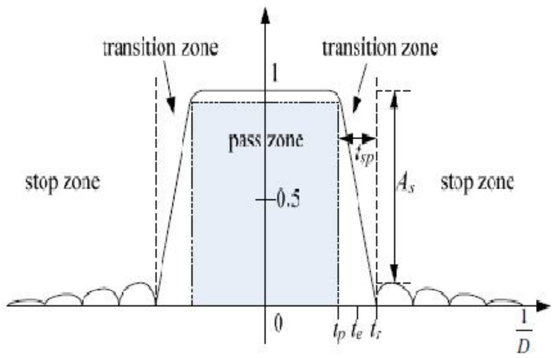


Fig 4. Illustration of time-domain window and CIR replication.

In order to derive an FDI suitable for the CE in SFN, we have to break through the limitation in (8). From Fig. 4, we can see the aliasing between the CIR and its replicas does not appear as long as the maximum delay τ_{max} < Tu / D. A possible solution is derived through shifting the symmetrical window to the right by half of the window width. The window shifting is implemented equivalently in the frequency domain by the phase rotation of the coefficient b[n], which translates a real FDI to a complex one. For an ideal CC-FDI, the original CIR can be separated even though τmax is extended to Tu / D, thus the CFR of data subcarriers can be accurately estimated. The first step is to design a RC-FDI of which the corresponding time-window width is τ_{max} / Tu; the second step is to transform the RC-FDI into the required CC-FDI. In the first step, if τmax is unknown, we can consider τ_{max} = NgTs for the worst case. Since the FDI can be viewed as a filtering process, the optimized FDI coefficient is obtained by meansof the existing methods of the filter design.In this paper the SLKW (sinc low pass with Kaiser window)is chosen.



Calculate the RC-LPF coefficient br[n] as (9)

$$b_r[n] = w[n] \cdot \frac{\sin(2\pi t_e n)}{\pi n} \quad (9)$$

By using br[n], in the second step we have the corresponding complex coefficient

$$b[n] = D \cdot b_r[n] e^{-i2\pi t_p n} \quad (10)$$

Drawback of this scheme is although the SLKW is simple and flexible; it does not always provide the maximum interpolator length, which is determined by the transition width and the side lobe suppression.

V. CONCLUSION

In this paper, we have given a historical review of popular multiplexing technique OFDM. We have also reported discussion, benefits, drawbacks and improvement in channel estimation algorithm in recent work includes, first paper A which shows better performance in presence of frequency offset and phase noise. Second paper B focused on the computation complexity and lowered it in comparison to previous algorithm. Third paper C contain algorithm for channel estimation with low computation overhead to cancel the impulsive noise. Last paper D which proposed algorithm for channel estimation in last year for pilot aided OFDM system and simulated in DVB-T/H receiver.

REFERENCES

[1] Adarsh B.Narasimhamurthy, Mahesh K.Banavar and Cihan Tepedelenliogu,―OFDM system for wireless communications, 2010.

[2] S. Weinstein and P. Ebert,-Data Transmission by Frequency Division Multiplexing Using the Discrete Fourier Transform IEEE Trans. vol.19, Issue: 5, pp. 628–634, Oct.1971.

[3] Ramjee Prasad, ―OFDM towards fixed and mobile broadband wirless access‖, Artech House, 2007.

[4] Ramjee Prasad, ―OFDM for wireless communication system‖,Artech House, 2004.

[5] Theodore S. Rappaport, ―Wireless Communications Principles and Practice‖, 2nd ed. Dec 31, 2001.

[6] Songping wu and Yeheskel bar-ness "OFDM channel estimation in the presence of frequency offset and phase noise"IEEE 2003.

[7] F. Daffara and O. Adami, "A new frequency detector for orthogonal multicarrier transmission techniques," IEEE 45th Vehicular Technology Conf., Chicago, IL, pp. 804-809, Jul.1995.

[8] X. Ma, G. B. Giannakis and S. Barbarossa, "Non data-aided frequency-offset and channel estimation in OFDM and related block transmissions," ICC' 2001, pp. 1866-1870, Helsinki, Finland, Jun. 2001.

[9] Xianbin Wang, Yiyan Wu and Jean-Yves Chouinard "Modified Channel Estimation Algorithms for OFDM Systems with Reduced Complexity" IEEE 2004.

[10] O.Edfors, M. Sandell,J.J van de beek,S.K.Wilson and P.O.Bar-jesson, "OFDM channel estimation by singular value decomposition," IEEE Trans Commun, vol. 46,pp.931-939, july 1998.

[11] Xuing Yu,ping Lin, Zhiqiang He and Weilig Wu "OFDM channel estimation with Impulse noise cancellation" IEEE 2007.

[12] Garcia, M.J.F.-G. and Rojo-Alvarez, J.L., "Support vector machines for robust channel estimation in OFDM", IEEE Signal Processing Letters, vol. 13, No.7, pp.397-400, Jul. 2006. [13]Guanghui Liu, Liaoyuan Zeng, Hongliang Li, and Zhengning Wang "Complex Coefficient Interpolation Based Channel Estimation for OFDM in Single –Frequency Networks"IEEE 2012

[14] Q. Yang, and K. S. Kwak, "Time-varying multipath channel estimation with superimposed training in CP-OFDM systems," ETRI Journal, vol. 28, no. 6, pp. 822–825, Dec. 2006.

DEVELOPMENT OF DISTANCE RELAYS USING MATLAB

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Abstract- Numerical relays are the result of the application of microprocessor technology in relay industry. Numerical relays have the ability to communicate with its peers, are economical and are easy to operate, adjust and repair. Modeling of numerical relay is important to adjust and settle protection equipment in electrical facilities and to train protection personnel. This paper describes distance relay model built using MATLAB environment. Data generated by MATLAB describes the voltage and current signals at the relay location both immediately before and during the fault. The signals include the dc offset and the high frequency traveling waves. The data is applied to the relay simulator, which then evaluates whether the impedance trajectory of the fault enters one or more of the operating zones. The results are presented in graphical form using an R-X diagram. The model is then verified by checking the model impedance measurement at different fault locations, and resistive faults. The paper demonstrates the benefits achieved when using a computer simulation of a relay in conjunction with a transient power system simulation.

Index Term- *Numerical distance relays; Impedance trajetory; Simulation; Fourier transform; MATLAB;*

I. INTRODUCTION

When a short-circuit fault occurs on a transmission line, the distance relays protecting the line trip the circuit breaker at either end of the line and disconnect the line from the network. To study the behavior of a distance relay during a short-circuit fault it is necessary to accurately simulate the distance relay and then inject into the simulator, the voltage and current signals seen by the relay during the fault. The voltage and current signals can be obtained from power system transmission line model in MATLAB.

The impedance seen by the distance relay during the fault. The evaluation of the operating performance of a distance relay using a dynamic simulator, can help determine the appropriate values for the zone one,zone two,zone three impedance settings and help determine whether an actual distance relay behaved correctly or incorrectly during a fault on the real network.The distance relays were modeled using MATLAB and the algorithm is based on the calculation of the voltage and current phasors using Fourier filters. At present the simulator include mho distance relay.

II. DEVELOPMENT OF DISTANCE RELAY

Distance relays used to calculate line impedance by measuring voltages and currents on one single end. The relays compare the setting impedance with the measured impedance to determine if the fault is inside or outside the protected zone.

They immediately release a trip signal when the impedance value is inside the zone 1 impedance circle of distance relay. For security protection consideration, the confirmation of a fault occurrence will not be made until successive trip signals are - released in one zone. Different formulas should be adopted when calculating the fault impedance due to different fault types. Table (1) indicates calculation formula for all of the fault types. Any three-phase faults can be detected from every formula in table (1). In order to reduce calculation burden, we design a fault detector and fault type selector. The fault detector can judge which fault type it is and then calculate fault impedance by selecting a suitable formula from table (1). If fault type judgment is not invoked first, then the impedance calculation by using the all six formulas shown in table (1) would be initiated simultaneously which causes much calculation burden.

III. DISTANCE RELAY OPERATING PRINCIPLES

When the distance relays receives voltage current signal, it converts it to a phasor. However faults on transmission lines cause the voltage and currents signals to be severely distorted.These signals may contain decaying dc components, subsystem frequency transients, high frequency oscillation quantities and etc.

The higher frequency components can be eliminated using low pass anti-aliasing filters with appropriate cut-off frequency, but the anti-aliasing filters cannot remove dc components and reject low frequency components. This makes the phasors very difficult to be quickiy estimated and affects the performance of digital relaying. Therefore, the Discrete Fourier Transform is usually used to remove the dc-offset components.DFT is a digital filtering algorithm that computes the magnitude and phase at discrete frequencies of a discrete time sequence

IV. SIMULATION PROCEDURE

When the distance relays receive discrete voltage and current signal, it converts it to a phasor. The Discrete Fourier Transform (DFT) is the most popular method to estimate fundamental phasors for digital relaying.

Table 1: Input signals of ground and phase distance relays

Distance Element	Voltage signal	Current signal
Phase A	V_a	$I_a + 3K_0I_0$
Phase B	V_b	$I_b + 3K_0I_0$
Phase C	V_c	$I_c + 3K_0I_0$
Phase A - Phase B	$V_a - V_b$	$I_a - I_b$
Phase B - Phase C	$V_b - V_c$	$I_b - I_c$
Phase C - Phase A	$V_c - V_a$	$I_c - I_a$

When a fault occurs on transmission lines, the voltage and current signals are severely distorted. These signals may contain decaying dc components, subsystem frequency transients, high frequency oscillation quantities, and etc. The higher frequency components can be eliminated using low pass anti-aliasing filters with appropriate cut-off frequency, but the anti-aliasing filters cannot remove decaying dc components and reject low frequency components.
This makes the phasors very difficult to be quickly estimated and affects the performance of digital relaying. Therefore, the Discrete Fourier transform is usually used to remove the dc-offset components.

The voltage and current data are derived using an MATLAB model of the power system. These data are discovered by using Data Acquisition System in MATLAB. The sampling rate used in the distance relay is 1.0 kHz. i.e (20 sample/cycle). The samples are 1ms a part.

In Data Acquisition System, the data input to digital filter "low pass filter" using cut-off frequency 360Hz this to remove the effects on the voltage and current signals of the traveling waves instigated by the fault. The data then be input to Discrete Fourier Transformer DFT window. The DFT is ideal method of detecting the fundamental frequency component in a fault signal. However, DFT, Least Error Square LES and Walsh Function algorithms are among the most popular phasor estimation techniques employed in numerical relays. As result, the magnitude and the phase angle of voltage and current obtained for the input signal, where it is then transformed to rectangular form.

The model is then proceeded to calculate the value of resistance and reactance of the line as seen by the relay by using the equations (3) and (4) . Figure 1, shows block diagram for distance relay modeling procedure. MATLAB program was used then to plot on graph characteristic of mho distance relay the behavior of Z during the sampling period.

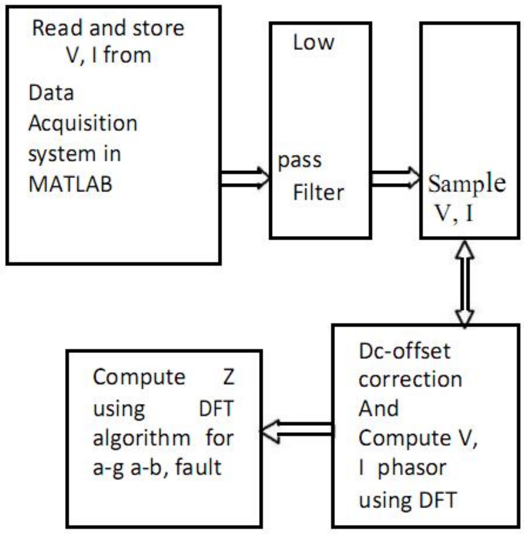

Figure .1, Procedure for development distance relay model.

V. DISCRETE FOURIER TRANSFORMER ALGORITHMS (DFT)

When a power system is operating under steady-state conditions, both the voltage and the current signals are periodic and the fundamental frequency component of each is at the power frequency. Therefore it is possible to calculate the impedance corresponding to a given voltage and current by determining the fundamental components of voltage and current using a discrete Fourier transform (DFT) technique.

Assuming the N samples are obtained for each period and that discrete time signals are X (K) then sampled signals are given as in equation (1).

$$X(n) = \sum_{k=0}^{N-1} X(k).e^{-i\frac{2\Pi nk}{N}} \quad \ldots\ldots\ldots\ldots(1)$$

$$X_1 = \frac{2}{\sqrt{2.N}} \sum_{k=0}^{N-1} X_k .e^{\frac{2\Pi nk}{N}} \quad \ldots\ldots\ldots\ldots(2)$$

Where n is the order of the harmonics. The fundamental frequency signals are the ones with n=1.k is the number of samples contained in the data window. In equation (2) all the N samples are used in the calculation of the fundamental frequency signal, resulting in a full cycle Discrete Fourier Transform (DFT).

The extraction of fundamental frequency components of voltage and current signals via discrete Fourier transformer is used and then impedance calculation, resistance and reactance is calculated from voltages and currents samples (K) at relaying point as in (3) and(4).

$$R_K = \frac{V_K}{I_K + 3\,.\,\mathrm{Re}(K_0)\,.\,I_{0K}} \quad \ldots\ldots.. (3)$$

$$X_K = \frac{V_K}{I_K + 3\,.\,\mathrm{Im}(K_0)\,.\,I_{0K}} \quad \ldots\ldots.. (4)$$

is Zero compensation factor $k0 = [\frac{Z0}{Z1} - 1]$

VI. POWER SYSTEM PARAMETERS

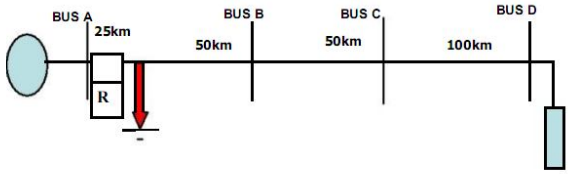

Figure 2: Single line diagram of simulated power network.

The current transformer ratio is 1000/1A and the voltage transformer ratio is 400kV/110V.
Setting of the relay is Zone-one = 46.52 ohm-secondary (80 percent of protected line1).
Zone-two = 90.38 ohm-secondary (100 percent of protected line1+ 50 percent of the line 2).
Zone-three = 129.52 ohm- secondary (100 percent of protected lines 1 and line 2 + 20 percent of line 3).
Given fig.3 Shows three zones of protection

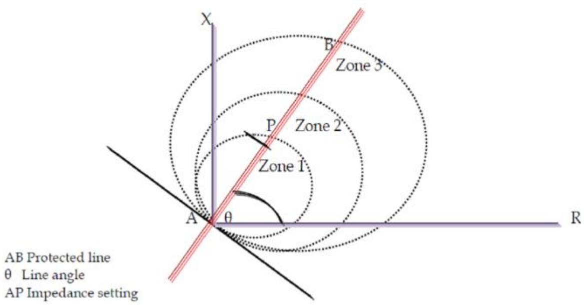

Fig.3.-Typical 3 zone protection

VII. SIMULATION RESULTS

The developed mho distance relay model is evaluated using data generated from Data Acquisition System. A single 400 kV over head line with length of 200 Km and a source with above source impedances is simulated. The overhead line is modeled as a distributed line model. Different fault locations on the transmission line with different fault resistances were simulated by MATLAB. The behavior of the mho distance relay model is as explained hereinafter.

A.Case one: - Single line to ground faults at different distances from the relay location.

Single line to ground faults were set on MATLAB model of the shown power system at 15 Km, 25 Km and 35 Km from the location of bus-A. The lengths are representing 10% to 80% of line A-B length,. Similarly few more Single line to ground faults at 5 Km, and 10 Km from the location of bus-B and bus-C were set. The voltage and current signal before and during fault were fed to the relay model. Figures 2 to 5, Show the impedance trajectory for these cases. In all cases the output results which are the impedance trajectory of the digital distance relay model had the expected behavior where the impedance trajectory calculations start the trajectory from the load area, before fault, and end up at the proper zone.

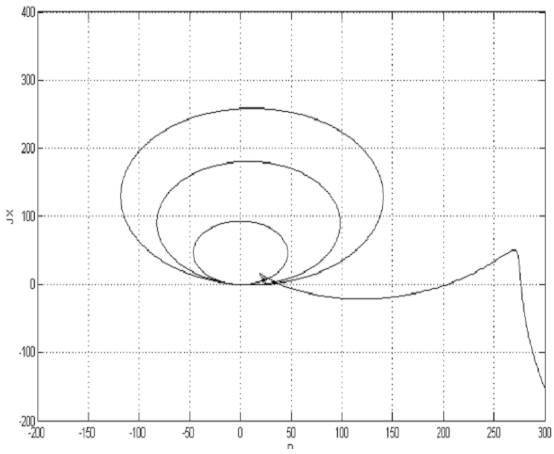

Fig.4. Trajectory of phase-to-ground fault at15km from bus A having Rf=6.8ohm

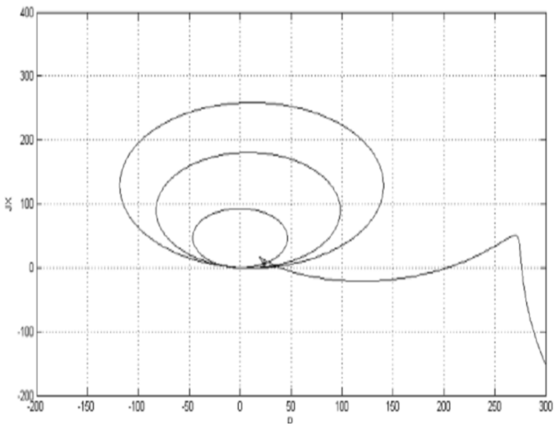

Fig.5. Trajectory of phase-to-ground fault at25km from bus A having Rf=6.8ohm

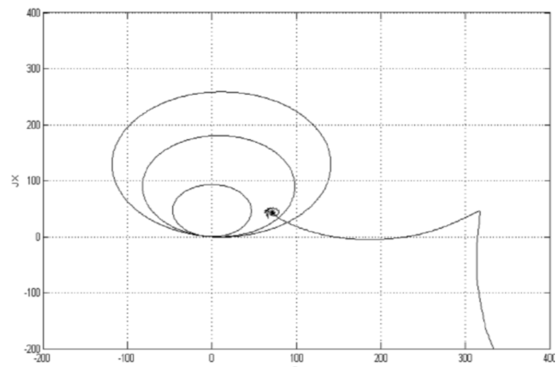

Fig.6. Trajectory of phase-to-ground fault at75km from bus A having Rf=6.8ohm

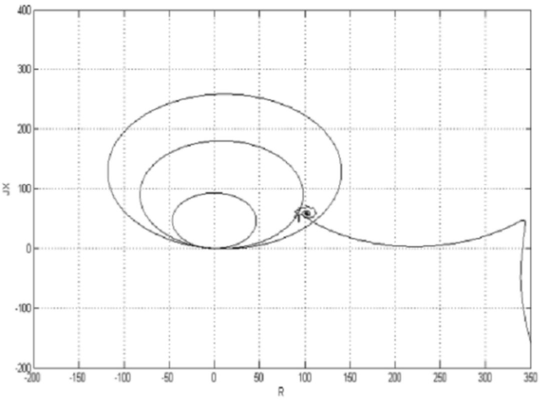

Fig.7. Trajectory of phase-to-ground fault at 120km from bus A having Rf=6.8o

B. Case two: Faults accompanied by a resistance

Single line to ground faults with different fault resistance were set on MATLAB model of the shown power system at different fault locations. Figures 7 to 9, Show the impedance trajectory for few of these cases. In all cases the output results which are the impedance trajectory of the digital distance relay model had the expected behavior where the effect of the fault resistance reflected on the value of the impedance seen by the relay. Impedance trajectory calculations start the trajectory from the load area, before fault, but, in few results, due to the resistance the relay misjudges the exact location of the fault.

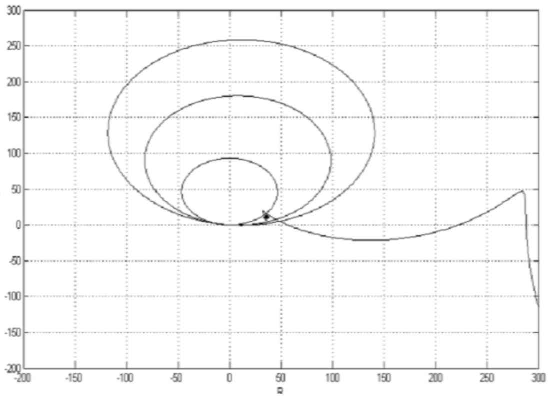

Fig.8. Trajectory of phase-to-ground fault at 35km from bus A having Rf=10

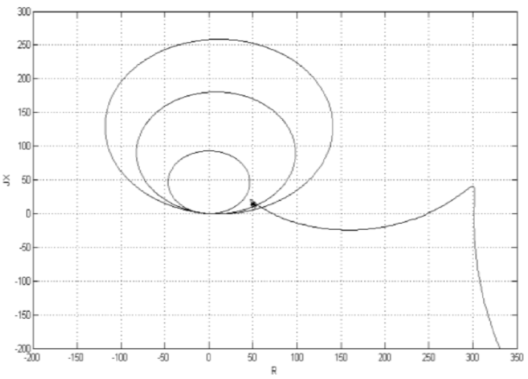

Fig.9. Trajectory of phase-to-ground fault at 35km from bus A having Rf=15

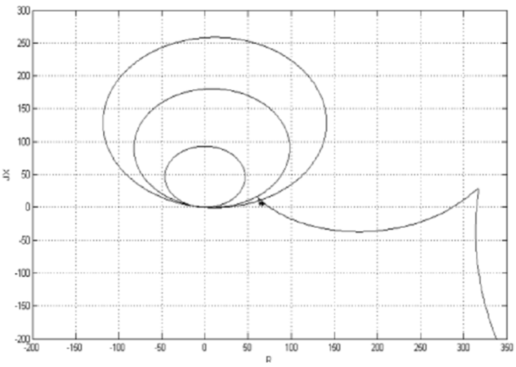

Fig.9. Trajectory of phase-to-ground fault at 35km from bus A having Rf=25

VIII. CONCLUSIONS

The impedance's trajectory after faults reflects numerical output of the impedance calculation. The output results reflected the behavior of the developed model under different fault locations and at different fault resistances. The simulation study presented in this paper assist in demonstrating the importance of and requirement for accurate dynamic modeling of distance protection relays. Different case studies have been presented in order to illustrate the response of the developed distance relay model at different operating scenarios, i.e. non-resistive faults and resistive faults. For the particular system studied it was found that the three-zone protection would not see a fault at the reach setting, resistive fault causes the relay to under-reach. The exact and misjudgment of the fault location in the cases demonstrated in this paper reflects the accuracy of the developed model. It is noted that the impedance's trajectory after faults were depends on digital filter type and relay algorithm. The distance relay model may be used as a training tool to help users understand how the relay works. The distance relay model offers an inexpensive alternative to evaluating a relay on a test set and generally will involve significantly less time and effort

REFERENCES

[1] Gerhard Ziegler “Numerical distance PROTECTION”, December 2005, second edition.

[2] A Phadake and J G Thorp,Computer Relaying for power system,John Wiley and Sons Inc,1990,ISBN

[3] T rin Saengsuwan,Peter Crossley "Simulation of Distance Relays for Protection Performance Evaluation", International symposium on electric power engineering, Stockholm power tech. Stockholm Sweden pp 491-496,June 94.

[4] “Power system Protection, Volume 4: Digital Protection and Signaling”, The Institute of Electrical Engineering, IEE, London 1995.

[5] The Output Processor (TOP),Copyright 1998-2001.

[6] MATLAB User’s guid,The Math Works Inc.,2006.

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