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A Fault Detection and Recovery Algorithm in Wireless Sensor Networks

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Abstract*—in the past few years wireless sensor networks have received a greater interest in application such as disaster management, border protection, combat field reconnaissance and security surveillance. Sensor nodes are expected to operate autonomously in unattended environments and potentially in large numbers. Failures are inevitable in wireless sensor networks due to inhospitable 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. Therefore, it is necessary that network failures are detected in advance and appropriate measures are taken to sustain network operation. In this paper we proposed a new mechanism to sustain network operation in the event of failure cause of energy-drained nodes. The proposed technique relies on the cluster members to recover the connectivity. The proposed recovery algorithm has been compared with some existing related work and proven to be more energy efficient [20].*

Key words: Sensor Networks, clustering, fault detection, fault recovery.

1. Introduction

Recent advances in MEMS (Micro-electro-mechanical systems) and wireless network technology have made the development of small, inexpensive, low power distributed devices, which are capable of local processing and wireless communication, a reality. Such devices are called sensor nodes. Sensors provide an easy solution to those applications that are based in the inhospitable and low maintenance areas where conventional approaches prove to be impossible and very costly. Sensors are generally equipped with limited data processing and communication capabilities and are usually deployed in an ad-hoc manner to in an area of interest to monitor events and gather data about the environment. Examples include environmental monitoring- which involves monitoring air soil and water, condition based maintenance, habitat monitoring, seismic detection, military surveillance, inventory tracking, smart spaces etc. Sensor nodes are typically disposable and expected to last until their energy drains. Therefore, it is vital to manage energy wisely in order to extend the life of the sensors for the duration of a particular task. [1-6]. Failures in sensor networks due to energy depletion are continuous and may increase. This often results in scenarios where a certain part of the network become energy constrained and stop operating after sometime. Sensor nodes failure may cause connectivity loss and in some cases network partitioning. In clustered networks, it creates holes in the network topology and disconnects the clusters, thereby causing data loss and connectivity loss [10]. Good numbers of fault tolerance solutions are available but they are limited at different levels. Existing approaches are based on hardware faults and consider hardware components malfunctioning only. Some assume that system software's are already fault tolerant as in [7, 8]. Some are solely focused on fault detection and do not provide any recovery mechanism [9]. Sensor network faults cannot be approached similarly as in traditional wired or wireless networks due to the following reasons [11]:

1. Traditional wired network protocol are not concerned with the energy consumptions as they are constantly powered and wireless ad hoc network are also rechargeable regularly.

2. Traditional network protocols aim to achieve point to point reliability, where as wireless sensor networks are more concerned with reliable event detection.

3. Faults occur more frequently in wireless sensor networks than traditional networks, where client machine, servers and routers are assumed to operate normally.

Therefore, it is important to identify failed nodes to guarantee network connectivity and avoid network partitioning. The New Algorithm recovery scheme is compared to Venkataraman algorithm proposed in [10]. The Venkataraman algorithm is the latest approach towards fault detection and recovery in wireless sensor networks and proven to be more efficient than some existing related work. It solely focused on nodes notifying its neighboring nodes to initiate the recovery mechanism. It can be observed from the simulation results that failure detection and recovery in our proposed algorithm is more energy efficient and quicker than that of Venkataraman algorithm. In [10], it has been found that Venkataraman algorithm is more energy efficient in comparison with Gupta and Younis [15].

Therefore, we conclude that our proposed algorithm is also more efficient than Gupta and Crash fault detection algorithm in term of fault detection and recovery.

This paper is organized as follows: Section 2 provides a brief review of related work in the literature. . In section 3, we provided a detail description of our clustering algorithm.

In section 4, we provided a detail description of our proposed solution. The performance evaluation of our proposed algorithm can be found in Section 5, Finally, section 6 concludes the paper.

2. Related Work

In this section we will give an overview about existing fault detection and recovery approaches in wireless sensor networks. A survey on fault tolerance in wireless sensor networks can be found in [12]. A detailed description on fault detection and recovery is available at [11]. WinMS [13] provides a centralized fault management approach. It uses the central manager with global view of the network to continually analyses network states and executes corrective and preventive management actions according to management policies predefined by human managers. The central manager detects and localized fault by analyzing anomalies in sensor network models. The central manager analyses the collected topology map and the energy map information to detect faults and link qualities. WinMS is a centralized approach and approach is suitable for certain application. However, it is composed of various limitations. It is not scalable and cannot be used for large networks. Also, due to centralize mechanism all the traffic is directed to and from the central point. This creates communication overhead and quick energy depletions. Neighboring co-ordination is another approach to detect faulty nodes. . For Example, the algorithm proposed for faulty sensor identification in [16] is based on neighboring co-ordination. In this scheme, the reading of a sensor is compared with its neighboring' median reading, if the resulting difference is large or large but negative then the sensor is very likely to be faulty. Chihfan et. al [17] developed a Self monitoring fault detection model on the bases of accuracy. This scheme does not support network dynamics and required to be pre configured. In [14], fault tolerance management architecture has been proposed called MANNA (Management architecture for wireless sensor networks). This approach is used for fault diagnosis using management architecture, termed as MANNA. This scheme creates a manager located externally to the wireless sensor network and has a global vision of the network and can perform complex operations that would not be possible inside the network. However, this scheme performs centralized diagnosis and requires an external manager. Also, the communication between nodes and the manager is too expensive for WSNs. In Crash fault detection scheme [18], an initiator starts fault detection mechanism by gathering information of its neighbors to access the neighborhood and this process continue until all the faulty nodes are identified. Gathering neighboring nodes information consumes significant energy and time consuming. It does not perform recovery in terms of failure. Gupta algorithm [15] proposed a method to recover from a gateway fault. It incorporates two types of nodes: gateway nodes which are less energy constrained nodes (cluster headers) and sensor nodes which are energy constrained. The less energy constrained gateway nodes maintain the state of sensors as well as multi-hop route for collecting sensors. The disadvantage is that since the gateway nodes are less energy constraint and static than the rest of the network nodes and they are also fixed for the life of the network. Therefore sensor nodes close to the gateway node die quickly while creating holes near gateway nodes and decrease network connectivity.

Also, when a gateway node die, the cluster is dissolved and all its nodes are reallocated to other healthy gateways. This consume more time as all the cluster members are involved in the recovery process. Venkataraman algorithm [10], proposed a failure detection and recovery mechanism due to energy exhaustion. It focused on node notifying its neighboring nodes before it completely shut down due to energy exhaustion. They proposed four types of failure mechanism depending on the type of node in the cluster. The nodes in the cluster are classified into four types, boundary node, pre-boundary node, internal node and the cluster head. Boundary nodes do not require any recovery but pre-boundary node, internal node and the cluster head have to take appropriate actions to connect the cluster. Usually, if node energy becomes below a threshold value, it will send a fail_report_msg to its parent and children. This will initiate the failure recovery procedure so that failing node parent and children remain connected to the cluster.

3. Cluster formation

The sensor nodes are dispersed over a terrain and are assumed to be active nodes during clustering.

- *Problem Definition*

The clustering strategy limits the admissible degree, *D* and the number of nodes in each cluster, *S*. The clustering aims to associate every node with one cluster. Every node does not violate the admissible degree constraint, *D* and every cluster does not violate the size constraint, *S* while forming the cluster. The number of clusters(*C*) in the network is restricted to a minimum of N/S, $N < C < N/S$, where *N* is the number of nodes in the terrain.

- *Sensor Network model*

A set of sensors are deployed in a square terrain. The nodes possesses the following properties

i. The sensor nodes are stationary.

ii. The sensor nodes have a sensing range and a transmission range. The sensing range can be related to the transmission range, Rt> 2rs.

iii. Two nodes communicate with each other directly if they are within the transmission range

iv. The sensor nodes are assumed to be homogeneous i.e. they have the same processing power and initial energy.

v. The sensor nodes are assumed to use different power levels to communicate within and across clusters.

vi. The sensor nodes are assumed to know their location and the limits *S* and *D*.

- *Description of the Clustering Algorithm*

Initially a set of sensor nodes are dispersed in the terrain. We assume that sensor nodes know their location and the limits *S* and *D*. Algorithms for estimating geographic or logical coordinates have been explored at length in the sensor network research [21, 22].

In our algorithm, the first step is to calculate *Eth* and *Eic* for every node i, N <I< 1. Eth is the energy spent to communicate with the farthest next hop neighbor. *Eic* is the total energy spent on each link of its next hop neighbors. Every node *i* has an initial energy, E_{init}. A flag bit called "*covered flag"* is used to denote whether the node is a member of any cluster or not. It is set to 0 for each node initially.

- *Calculation of Eth and Eic*

i. Nodes send a message *hello_msg* along with their coordinates which are received by nodes within the transmission range. For example in figure (1) nodes *a*,*b*,*c*,*d*,*w*,*x*,*y* are within transmission range of *v*.

ii. After receiving the *hello_msg*, the node *v* calculates the distance between itself and nodes *a*,*b*,*c*,*d*,*w*,*x*,*y* using the coordinates from *hello_msg*. It stores the distance d_i and the locations in the *dist_table*.

iii. Nodes within the sensing range are the neighbours of a node. In figure (1) nodes *w*,*x*,*y*,*b* are neighbours of *v*.

iv. Among the nodes within the sensing range, it chooses the first *D* closest neighbours as its potential candidates for next hop. Assuming *D*=3, in figure (1), the closest neighbours of *v* are *w*,*x*,*y*. v. Among the potential candidates, the farthest node's distance, d_{max} is taken for the calculation of E_{th} .

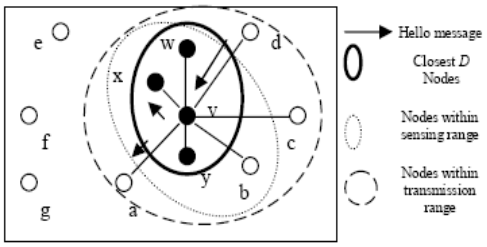

Figure 1. Topology

vi. Suppose a node needs power E to transmit a message to another node who is at a distance 'd' away, we use the formula $E = E=kd^{c}$ [7,8], where k and c are constants for a specific wireless system. Usually 2<c<4. In our algorithm we assume k=1, c=2. For a node v, d *1*d^2max = E thv, since there are D members to which a node sends message.

vii. *Eic* is the total energy spent on each of link of the *D*closest neighbours. For a node *v*,

$$E_{ic\,v} = \sum_{i=1}^{D} d_{iv}^2$$

where k= 1. d_{iv} is the distance between node *i* and node *v*. After the calculation of threshold energy *Eth*, nodes become eligible for cluster head position based on their energies. A node *v* becomes eligible for the cluster head position if its $E_{init} > Eth_v$. *and*. node with the second $E_{init} > Eth_v$ becomes secondary Cluster heed.. When no nodes satisfy this condition or when there is insufficient number of cluster heads, the admissible degree *D* is reduced by one and then *Eth* is recalculated. The lowest value that *D* can reach is one. In a case where the condition $E_{init} > Eth_v$ is never satisfied at all, clustering is not possible because no node can support nodes other than itself. There may also be situations where all the nodes or more number of nodes are eligible for being cluster heads. A method has been devised by which the excess cluster heads are made to relinquish their position.

- *Relinquishing of the cluster head position*

i. Every cluster head sends a message *cluster_head_status msg* and *Eic* to its neighbours (within sensing range).

ii. Every cluster head keeps a list of its neighbor cluster heads along with its *Eic*

iii. The nodes which receive *Eic* lesser than itself relinquishes its position as a cluster head.

iv. The cluster heads which are active send their messages to the cluster_head_manager outside the network. Thecluster_head_manager has the information of the desired cluster head count.

v. If the number of cluster heads are still much more than expected, then another round of cluster head relinquishing starts. This time the area covered would be greater than sensing range.

vi. The area covered for cluster head relinquishing keeps increasing till the desired count is reached.

- *Choosing cluster members*

i. The cluster head select the closest *D* neighbours as next hop and sends them the message *cluster_join_msg*. The *cluster_join_msg* consists of cluster ID, S_a, *D*, *S*, *covered flag*. S_a is (*S*-1)/number of next hop members

ii. Energy is expended when messages are sent. This energy, *Eic* is calculated and reduced from the cluster head's energy.

iii. The cluster head's residual energy $E_r = E_{init} - Eic$. E_{init} is the initial energy when the cluster is formed by the cluster head.

iv. After receiving the *cluster_join_msg*, the nodes send a message, *cluster_join_confirm_msg* to the cluster head if they are uncovered, else they send a message, *cluster_join_reject_msg*.

v. After a *cluster_join_confirm_msg*, they set their *covered_flag* to 1.

vi. The next hop nodes now select *D*-1 members as their next hop members. *D*-1 members are selected because they are already associated with the node which selected them. For example, in figure (1) where *D* =3, a node *v* selects node *w*, node *x* and node *y*. In the next stage, node y selects

only node *a* and node *b* because it is already connected to node *v* making the *D* = 3.

vii. After selecting the next hop members, the residual energy is calculated , E_r (new) = E_r (old) - *Eic*

viii. This proceeds until S is reached or until all nodes have their *covered_flag* set to 1.

- *Tracking of the size*

The size *S* of the cluster is tracked by each and every node. The cluster head accounts for itself and equally distributes *S*-1 among its next hop neighbors by sending a message to each one of them. The neighbours that receive the message account for themselves and distribute the remaining among all their neighbours except the parent. The messages propagate until they reach a stage where the size is exhausted. If the size is not satisfied, then the algorithms terminates if all the nodes have been covered. After the cluster formation, the cluster is ready for operation. The nodes communicate with each other for the period of network operation time.

4. Cluster Heed Failure recovery Algorithm

We employ a backup secondary cluster heed which will replace the cluster heed in case of failure, no further messages are required to send to other cluster members to inform them about the new cluster heed. Cluster heed and secondary cluster heed are known to their cluster members. If cluster heed energy drops below the threshold value, it then sends a message to its cluster member including secondary cluster heed. Which is an indication for secondary cluster heed to standup as a new cluster heed and the existing cell manager becomes common node and goes to a low computational mode. Common nodes will automatically start treating the secondary cluster heed as their new cluster heed and the new cluster heed upon receiving updates from its cluster members; choose a new secondary cluster heed. Recovery from cluster heeds failure involved in invoking a backup node to standup as a new cluster heed.

5. Performance Evaluation

The energy model used is a simple model shown in [19] for the radio hardware energy dissipation where the transmitter dissipates energy to run the radio electronics and the power amplifier, and the receiver dissipates energy to run the radio electronics. In the simple radio model [19], the radio dissipates E_{elec} = 50 nJ/bit to run the transmitter or receiver circuitry and E_{amp} = 100 (pJ/bit)/m^2 for the transmit amplifier to achieve an acceptable signal-to-noise ratio. We use MATLAB Software as the simulation platform, a high performance discrete-event Java-based simulation engine that runs over a standard Java virtual machine.

The simulation parameters are explained in Table 1.

We compared our work with that of Venkataraman algorithm [10], which is based on recovery due to energy exhaustion.

Table 1.Simulation parameters

Simulation parameters	Value
terrain dimensions	1 km^2
total number of nodes in terrain, *N*	100–1200
transmission range	100–450 m
cluster size limit, *S*	10–50
supportable degree, *D*	3–10

In Venkataraman algorithm, nodes in the cluster are classified into four types: boundary node, pre-boundary node, internal node and the cluster head. Boundary nodes does not require any recovery but

pre-boundary node, internal node and the cluster head have to take appropriate actions to connect the cluster. Usually, if node energy becomes below a threshold value, it will send a fail_report_msg to its parent and children. This will initiate the failure recovery procedure so that failing node parent and children remain connected to the cluster. A join_request_mesg is sent by the healthy child of the failing node to its neighbors. All the neighbors with in the transmission range respond with a join_reply_mesg/join_reject_mesg messages. The healthy child of the failing node then selects a suitable parent by checking whether the neighbor is not one among the children of the failing node and whether the neighbor is also not a failing node. In our proposed mechanism, common nodes does not require any recovery but goes to low computational mode after informing their cell managers. In Venkataraman algorithm, cluster head failure cause its children to exchange energy messages. The children who are failing are not considered for the new cluster-head election. The healthy child with the maximum residual energy is selected as the new cluster head and sends a final_CH_mesg to its members. After the new cluster head is selected, the other children of the failing cluster head are attached to the new cluster head and the new cluster head becomes the parent for these children. This cluster head failure recovery procedure consumes more energy as it exchange energy messages to select the new cluster head. Also, if the child of the failing cluster head node is failing as well, then it also require appropriate steps to get connected to the cluster. This can abrupt network operation and is time consuming.

In our proposed algorithm, we employ a back up secondary cluster heed which will replace the cluster heed in case of failure.

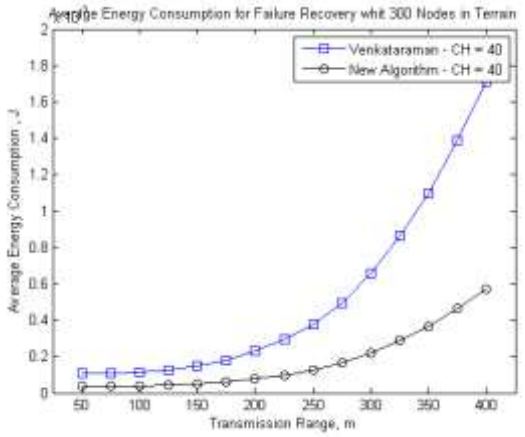

Fig. 2. Average time for cluster head recovery

no further messages are required to send to other cluster members to inform them about the new cluster heed Figs. 2 and 3 compare the average energy loss for the failure recovery of the three algorithms. It can be observed from Fig. 2 that when the transmission range increases, the greedy algorithm expends the maximum energy when compared with the Gupta algorithm and the proposed algorithm. However, in Fig. 3, it can be observed that the Gupta algorithm spends the maximum energy among the other algorithms when the number of nodes in the terrain increases.

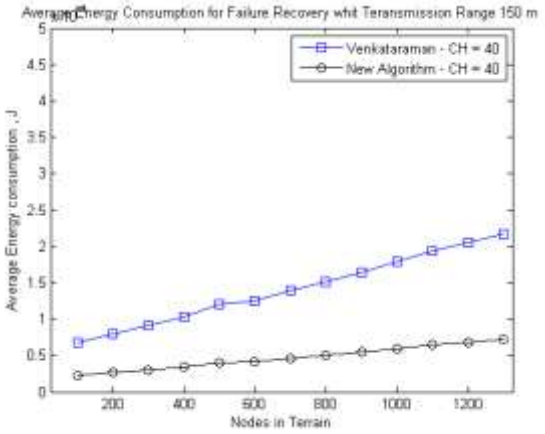

Fig. 3. Average time for cluster head recovery

6. Conclusion

In this paper, we have proposed a cluster-based recovery algorithm, which is energy-efficient and responsive to network topology changes due to sensor node failures. The proposed cluster-head failure-recovery mechanism recovers the connectivity of the cluster in almost less than of the time taken by the fault-tolerant clustering proposed by Venkataraman. The Venkataraman algorithm is the latest approach towards fault detection and recovery in wireless sensor networks and proven to be more efficient than some existing related work. Venkataraman algorithm is more energy efficient in comparison with Gupta and Algorithm Greedy Therefore, we conclude that our proposed algorithm is also more efficient than Gupta and Greedy[20] algorithm in term of fault recovery.

The faster response time of our algorithm ensures uninterrupted operation of the sensor networks and the energy efficiency contributes to a healthy lifetime for the prolonged operation of the sensor network.

References

[1] A. Bharathidasas, and V. Anand, “Sensor networks: An overview”, Technical report, Dept. of Computer Science, University of California at Davis, 2002

[2] D. Estrin, R. Govindan, J. Heidemann, and S. Kumar, "Next century challenges: Scalable coordination in sensor networks", in Proceedings of ACM Mobicom, Seattle, Washington, USA, August 1999, pp. 263-- 270, ACM.

[3] I.F. Akyildiz, W. Su, Y. Sankarasubramaniam and E. Cayirci, "A Survey on Sensor Networks", IEEE Communications Magazine, pp. 102--114, August 2002.

[4] D. Estrin, L. Girod, G. Pottie, M. Srivastava, “Instrumenting the world with wireless sensor networks”, In Proceedings of the International Conference on Acoustics, Speech and Signal Processing (ICASSP 2001.

[5] E. S. Biagioni and G. Sasaki, “Wireless sensor placement for reliable and efficient data collection”, in the 36th

International Conference on Systems Sciences, Hawaii, January 2003.
[6] G. Gupta and M. Younis, "Load-Balanced Clustering in Wireless Sensor Networks", in the Proceedings of International Conference on Communication (ICC 2003), Anchorage, AK, May 2003.
[7] J. Chen, S. Kher and A. Somani, "Distributed Fault Detection of Wireless Sensor Networks", in DIWANS'06. 2006. Los Angeles, USA: ACM Pres.
[8] F. Koushanfar, M. Potkonjak, A. Sangiovanni- Vincentelli, "Fault Tolerance in Wireless Ad-hoc Sensor Networks", Proceedings of IEEE Sensors 2002, June, 2002.
[9] W. L. Lee, A. Datta, and R. Cardell-Oliver, "Network Management in Wireless Sensor Networks", to appear in Handbook on Mobile Ad Hoc and Pervasive Communications, edited by M. K. Denko and L. T. Yang, American Scientific Publishers.
[10] G. Venkataraman, S. Emmanuel and S.Thambipillai, "Energy-efficient cluster-based scheme for failure management in sensor networks" IET Commun, Volume 2, Issue 4, April 2008 Page(s):528 – 537
[11] L. Paradis and Q. Han, "A Survey of Fault Management in Wireless Sensor Networks", Journal of Network and Systems Management, vol. 15, no. 2, pp. 171-190, 2007.
[12] L. M. S. D. Souza, H. Vogt and M. Beigl, "A survey on fault tolerance in wireless sensor networks", 2007.
[13] W. L Lee, A.D., R. Cordell-Oliver, WinMS: Wireless Sensor Network-Management System, An Adaptive Policy-Based Management for Wireless Sensor Networks. 2006.
[14] L. B. Ruiz, I. G.Siqueira, L. B. Oliveira, H. C. Wong, J.M. S. Nigeria, and A. A. F. Loureiro. "Fault management in event-driven wireless sensor networks", MSWiM'04, October 4-6, 2004, Venezia, Italy
[15] G. Gupta and M. Younis; Fault tolerant clustering of wireless sensor networks; WCNC'03, pp. 1579.1584.
[16] M. Ding, D. Chen, K. Xing, and X. Cheng, "Localized fault-tolerant event boundary detection in sensor networks", in Proceedings of the 24th Annual Joint Conference of the IEEE Computer and Communications Societies (INFOCOM '05), vol. 2, pp. 902–913, Miami, Fla, USA, March 2005
[17] C. Hsin and M.Liu, "Self-monitoring of Wireless Sensor Networks", Computer Communications, 2005. 29: p. 462-478
[18] S. Chessa and P. Santi, "Crash faults identification in wireless sensor networks", Comput. Commun., 2002, 25, (14), pp. 1273-1282.
[19] W. R. Heinzelman, A. Chandrakasan, and H. Balakrishnan, "Energy-Efficient Communication Protocol for Wireless Microsensor Networks," Proc. Hawaii Int'l Conf. System Sciences 2000.
[20] GUPTA G., YOUNIS M.: 'Fault-tolerant clustering of wireless sensor networks'. Proc. IEEE WCNC, New Orleans, USA,March 2003, vol. 3, p. 1579 – 1584
[21] N. Bulusu, J. Heidemann and D. Estrin, "GPS-less Low Cost Outdoor Localization For Very Small Devices", IEEE Personal Communications, Special Issue on "Smart Spaces and Environments", Vol. 7, No. 5, pp. 28-34, October 2000.
[22] Radhika Nagpal, "Organizing a Global Coordinate System from Local Information on an Amorphous Computer", MIT AI Memo 1666, August 1999

Bacterial Foraging Optimization Based Load Frequency Control of Interconnected Power Systems with Static Synchronous Series Compensator

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Abstract: This paper proposes a design of Bacterial Foraging Optimization (BFO) based optimal integral controller for the load-frequency control of two-area interconnected thermal reheat power systems without and with Static Synchronous Series Compensator (SSSC) in the Tie-line. The BFO Technique is used to optimize the optimal integral gain setting by minimizing quadratic performance index. The main application of SSSC is to stabilize the frequency oscillations of the inter area mode in the interconnected power system by the dynamic control of tie-line power flow. Simulation studies reveal that with SSSC units, the deviations in area frequencies and inter area tie-line power are considerably improved in terms of peak deviations and setting time as compared to the output responses of the system obtained without SSSC units.

Keywords: Bacterial Foraging Optimization, Integral Controller, Integral Square Error Criterion, Static Synchronous Series Compensator, Load-Frequency Control.

1. Introduction

Power systems, with the increase in size and complexity, require interconnection between the systems to ensure more reliable power supply even under emergencies by sharing the spinning reserve capacities. In this aspect, the Load-Frequency Control (LFC) and inter- area tie-line power flow control, a decentralized control scheme is essential. The paper proposes a control scheme that ensures reliability and quality of power supply, with minimum transient deviations and ensures zero steady state error. The importance of decentralized controllers for multi area load-frequency control system, where in, each area controller uses only the local states for feedback, is well known. The stabilization of frequency oscillations in an interconnected power system becomes challenging when implemented in the future competitive environment. So advanced economic, high efficiency and improved control schemes [1]-[6] are required to ensure the power system reliability. The conventional load-frequency controller may no longer be able to attenuate the large frequency oscillation due to the slow response of the governor [7]. The recent advances in power electronics have led to the development of the Flexible Alternating Current Transmission Systems (FACTS). These FACTS devices are capable of controlling the network condition in a very fast manner [6] and because of this reason the usage of FACTS devices are more apt to improve the stability of power system. SSSC can be installed in series with tie line between any interconnected areas, can be applied to stabilize the area frequency oscillations by high speed control of tie-line power through the interconnections. In addition it can also be expected that the high speed control of SSSC can be coordinated with slow speed control of governor system for enhancing stabilization of area frequency oscillations effectively [7]. A conventional lead/lag structure is preferred by the power system utilities because of the ease often on-line tuning and also lack of assurance of the stability by few adaptive or variable structure techniques. Nowadays power system complex are being solved with the use of Evolutionary Computation (EC) such as Differential Evolution (DE) [9], Genetic Algorithms [GAs], Practical Swarm Optimizations [PSO][10] Ant Colony Optimization[ACO][11], which are some of the heuristic techniques having immense capability of determining global optimum. Classical approach based optimization for controller gains is a trial and error method and extremely time consuming when several parameters have to be optimized simultaneously and provides suboptimal result. Some authors have applied genetic algorithm (GA) to optimize controller gains more effectively and efficiently than the classical approach. Recent research has brought out some deficiencies in GA performance [13], [14]. The premature convergence of GA degrades its search capability. The Bacterial Foraging Optimization [BFO] mimics how bacteria forage over a landscape of nutrients to perform parallel non gradient optimization [15]. The BFO algorithm is a computational intelligence based technique that is not large affected by the size and non-linearity of the problem and can be convergence to the optimal solution in many problems where most analytical methods faith convergence. A more recent and powerful evolutionary computational technique “Bacterial Foraging” (BF) [16] is found to be user friendly and is adopted for simultaneous optimization of several parameters for both primary and secondary control loops of the governor.

The simulation results show that the dynamic performance of the system is improved by using the proposed controller. The organizations of this paper are as follows. In section 2 problem formulation is described. In section 3 focuses on the design and implementation of SSSC unit. The output response of the system is investigated with the application of SSSC unit in section 4. Overview of BFO is

described in section 5.Section 6 presents the simulations and its results; finally a conclusion is discussed in section 7.

2. Problem Formulation

The state variable equation of the minimum realization model of ' N ' area interconnected power system may be expressed as [5]

$$\dot{x} = Ax + Bu + \Gamma d \quad (1)$$
$$y = Cx$$

where $x = [x_1^T, \Delta p_{ei} \ldots x_{(N-1)}^T, \Delta p_{e(N-1)} \ldots x_N^T]^T$,

n - state vector

$$n = \sum_{i=1}^{N} n_i + (N-1)$$

$u = [u_1, .. u_N]^T = [\Delta P_{C1} .. P_{CN}]^T$, N -Control input vector

$d = [d_1, .. d_N]^T = [\Delta P_{D1} .. P_{DN}]^T$, N - Disturbance input vector

$y = [y_1 .. y_N]^T$, $2N$ - Measurable output vector

where A is system matrix, B is the input distribution matrix, Γ is the disturbance distribution matrix, C is the control output distribution matrix, x is the state vector, u is the control vector and d is the disturbance vector consisting of load changes.

3. Application of SSSC for the Proposed Work

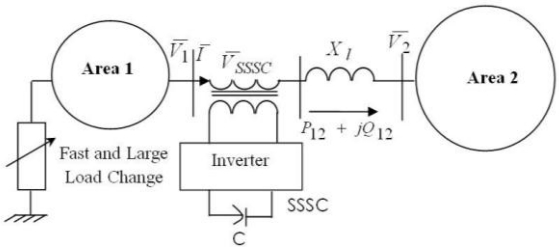

Figure 1. An SSSC in a two area interconnected power system

Figure.1 shows the two-area interconnected power system with a configuration of SSSC used for the proposed control design. It is assumed that a large load with rapid step load change has been experienced by area1.This load change causes serious frequency oscillations in the system. Under this situation, the governors in an area 1 cannot sufficiently provide adequate frequency control. On the other hand, the area 2 has large control capability enough to spare for other area. Therefore, an area 2 offers a service of frequency stabilization to area 1 using the SSSC. Since SSSC is a series connected device, the power flow control effect is independent of an installed location. In the proposed design method, the SSSC controller uses the frequency deviation of area 1 a local signal input. Therefore the SSSC is placed at the point near area1. Moreover the SSSC is utilized as the energy transfer device from area 2 to area1. As the frequency fluctuation in area 1 occurs, the SSSC will provide the dynamic control of the tie-line power by exploiting the system interconnections as the control channels and the frequency oscillation can be stabilized

3.1 Mathematical Model of the SSSC

In this study, the mathematical model of the SSSC for stabilization of frequency oscillations is derived from the characteristics of power flow control by SSSC [8]. By adjusting the output voltage of SSSC ($\bar{V}_{SSSC}$), the tie-line power flow (P_{12}+jQ_{12}), can be directly controlled as shown in fig1. Since the SSSC fundamentally controls only the reactive power, then the phasor $\bar{V}_{SSSC}$ is perpendicular to the phasor of line current $\bar{I}$, which can be expressed as

$$\bar{V}_{SSSC} = jV_{SSSC}\bar{I}/I \quad (2)$$

Where V_{SSSC} and I are magnitudes of $\bar{V}_{SSSC}$ and $\bar{I}$ respectively. Note That

Where $\bar{I}/I$ is a unit vector of line current. Therefore, the current $\bar{I}$ in fig 1, can be expressed as

$$\bar{I} = (\bar{V}_1 - \bar{V}_2 - jV_{sssc}\bar{I}/I)/jX_l \quad (3)$$

Where X_l is the reactance of the tie line, $\bar{V}_1$ and $\bar{V}_2$ are the bus voltages at bus 1 & 2 respectively. The active power and reactive power flow through bus 1 are

$$P_{12} + jQ_{12} = \bar{V}_1\bar{I}^* \quad (4)$$

Where $\bar{I}^*$ is conjugate of $\bar{I}$. Substituting $\bar{I}$ from (3) in (4),

$$P_{12} + jQ_{12} = \frac{V_1V_2}{X_l}\sin(\delta_1 - \delta_2) - V_{sssc}\frac{\bar{V}_1\bar{I}^*}{X_lI} + j\left(\frac{V_1^2}{X_l} - \frac{V_1V_2}{X_l}\cos(\delta_1 - \delta_2)\right) \quad (5)$$

Where $\bar{V}_1 = V_1e^{j\delta_1}$ and $\bar{V}_2 = V_2e^{j\delta_2}$.

From eqn (5) and (4), gives

$$P_{12} = \frac{V_1V_2}{X_l}\sin(\delta_1 - \delta_2) - \frac{P_{12}}{X_lI}V_{sssc} \quad (6)$$

The second term of right hand side of eqn(6) is the active power controlled by SSSC. Here, it is assumed that V_1 and V_2 are constant, and the initial value of V_{SSSC} is zero. i.e., V_{SSSC}=0. By linearzing (5) about an initial operating point

$$\Delta P_{12} = \frac{V_1V_2\cos(\delta_{10} - \delta_{20})}{X_l}(\Delta\delta_1 - \Delta\delta_2) - \frac{P_{120}}{X_lI_o}\Delta V_{sssc} \quad (7)$$

where subscript "0" denotes the value at the initial operating point by varying the SSSC output voltage ΔV_{SSSC} ,the power output of SSSC can be controlled as $\Delta P_{sssc} = (P_{120}/X_lI_0)\Delta V_{sssc}$. In equation (7) implies that the SSSC is capable of controlling the active power independently. In this study, the SSSC is represented by the power flow controller where the control effect of active power by SSSC is expressed by ΔP_{SSSC} instead of $(P_{120}/X_lI_0)\Delta V_{sssc}$. Eqn (7) can also be expressed as

$$\Delta P_{12} = \Delta P_{T12} + \Delta P_{sssc} \quad (7a)$$

Where $\Delta P_{T12} = \frac{V_1V_2\cos(\delta_{10} - \delta_{20})}{X_l}(\Delta\delta_1 - \Delta\delta_2)$

$$= T_{12}(\Delta\delta_1 - \Delta\delta_2) \quad (8)$$

Where T_{12} is the synchronizing power co-efficient

3.2 Structure of SSSC used as Damping Controller

The active power controller of SSSC has a structure of the Lead/Lag compensator with output signal $\Delta P_{ref.}$ In this study the dynamic characteristics of SSSC is modeled as the first order controller with time constant TSSSC. It is to be noted that the injected power deviations of SSSC, ΔPSSSC, acting positively on the area1 reacts negatively on the area2.

Therefore ΔPSSSC flow into both area with different signs (+,-) simultaneously.

The commonly used Lead-Lag structure is chosen in this study as SSSC based supplementary damping controller as shown in Fig2. The structure consists of a gain block. A signal washout block and Two-stage phase compensation block. The phase compensation block provides the appropriate phase-lead characteristics to compensate for the phase lag between input and output signals. The signals associated with oscillations in input signal to pass unchanged without it steady changes in input would modify the output the input signal of the proposed SSSC-based controller is frequency deviation Δf and the output is the change in control vector ΔPSSSC . From the view point of the washout function the value of washout time constant is not critical in Lead-Lag structured controllers and may be in the range 1 to 20seconds. From the view point of the washout function the value of washout time constant is not critical in Lead-Lag structured controllers and may be in the range 1 to 20seconds. In the present study, a washout time constant of Tw=10s is used. The controller gain K; and the time constant *T1*, *T2*, *T3* and *T4* are searched on the objective function using BFO Technique are determined

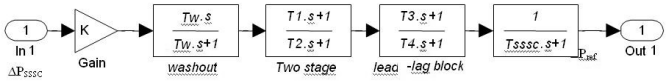

Figure 2. Structure of SSSC-based damping controller

4. System Investigated

Since the system under consideration is exposed to a small change in load during its normal operation, a linearized model is taken for the study.

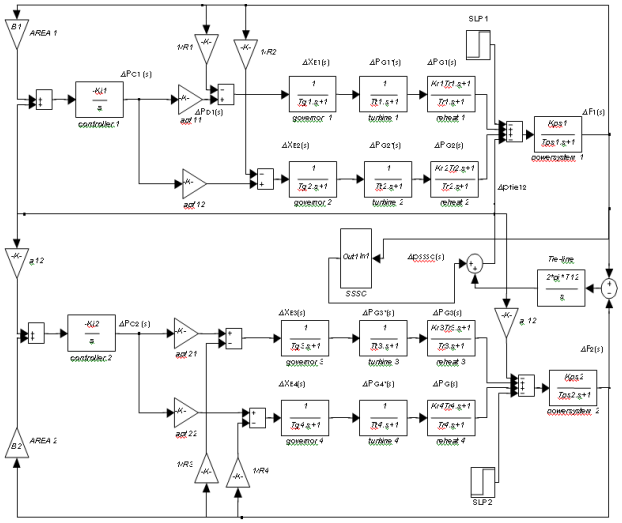

Figure:3 Transfer function model of tow area interconnected thermal re-heat power systems with SSSC unit

Investigations have been carried out in the two equal area interconnected thermal power system and each area consists of two reheat units as shown in Figure 3. The nominal parameters are given in Appendix. apf11 and apf12 are the ACE participation factors in area 1 and apf21 and apf22 are the ACE participation factor in area2. Note that apf11+apf12 = 1.0 and apf21+apf22 = 1.0. In this study the active power model of SSSC is fitted in the tie-line near area1 to examine its effect on the power system performance. The following objective function (9) is used to find the optimum gain of Integral controller using BFO technique for the two-area two unit reheat power system without and with SSSC unit. MATLAB version 7.01 has been used to obtain the dynamic responses for a step load perturbation of 1% in area1.

$$J = \int_0^T \{(\beta_1 \Delta f_1)^2 + (\beta_2 \Delta f_2)^2 + (\Delta P_{tie12})^2\} \quad (9)$$

5. Bacterial Foraging Optimization Technique

BFO method was invented by Kevin M. Possino [16] motivated by the natural selection which tends to eliminates the animals with poor foraging strategies and favor those having successful foraging strategies. The foraging strategy is governed by four processes namely chemotaxis, swarming, reproduction and elimination and dispersal

(1) Chemotaxis:

Chemotaxis process is the characteristics of movement of bacteria in search of food and consists of two processes namely swimming and tumbling .A bacterium is said to be swimming if it moves in a predefined direction, and tumbling if it starts moving in an altogether different direction. Let, j be the index of chemotactic step, k be reproduction step and *l*' be the elimination dispersal event. $\theta_i(j,\ k,\ l)$ is the position of i[th] bacteria at j[th] chemo tactic step k[th] reproduction step and l[th] elimination

$$\theta^i(j+1,\ k,\ l) = \theta^i(j,\ k,\ l) + C(i)\frac{\Delta(i)}{\sqrt{\Delta^T(i)\ \Delta(i)}} \quad (10)$$

Where $C(i)$ denotes step size

$\Delta(i)$ Random vector

$\Delta^T(i)$ Transpose of vector $\Delta(i)$

If the health of the bacteria improves after the tumble, the bacteria will continue to swim to the same direction for specified steps (or) until the health degrades

(2) Swarming :

Bacteria exhibits swarm behavior ie. Healthy bacteria try to attract other bacterium so that together they reach the desired location (solution point) more rapidly .The effect of swarming is to make the bacteria congregate into groups and moves as concentric patterns with high bacterial density Mathematically swarming behavior can be model .

$$J_{CC}(\theta,\ p(j,\ k,\ l)) = \sum_{i=1}^{S} J^i{}_{CC}(\theta,\ \theta_i(j,\ k,\ l))$$

$$= \sum_{i=1}^{S}\left[-d_{attract}\exp(-\omega_{attract})\sum_{m=1}^{p}(\theta^m-\theta_m^i)^2\right] + \sum_{i=1}^{S}\left[-h_{repelent}\exp(-\omega_{repelent})\sum_{m=1}^{p}(\theta^m-\theta_m^i)^2\right] \quad (11)$$

Where

J_{CC} - Relative distance of each bacterium from the fittest bacterium

S - Number of bacteria

p - Number of parameters to be optimized

θ^m - Position of the fittest bacteria

$d_{attract}$, $\omega_{attract}$, $h_{repelent}$, $\omega_{repelent}$ - parameters

(3) Reproduction:
In this step, population members who have had sufficient nutrients will reproduce and the least healthy bacteria will die. The healthier population replaces unhealthy bacteria which gets eliminated wing to their poorer foraging abilities. This makes the population of bacteria constant in the evolution *process.*

(4) Elimination and dispersal:
In the evolution process a sudden unforeseen event may drastically alter the evolution and may cause the elimination and or dispersion to a new environment .Elimination and dispersal helps in reducing the behavior of stagnation i.e, being trapped in a premature solution point or local optima.

5.1 Bacterial Foraging Algorithm

In case of BFO technique each bacterium is assigned with a set of variable to be optimized and are assigned with random values [Δ] within the universe of discourse defined through upper and lower limit between which the optimum value is likely to fall. In the proposed method integral gain KI_i(i=1,2) scheduling, each bacterium is allowed to take all possible values within the range and the cost objective function [J] which is represented by eqn (9) is minimized . In this study, the BFO algorithm reported in [16] is found to have better convergence characteristics and is implemented as follows.
Step 1- Initialization

a. Number of parameter (p) to be optimized. In this study K_{I1} and K_{I2}
b. Number of bacterial (S) to be used for searching the total region.
c. Swimming length (Ns), after which tumbling of bacteria will be undertaken in a chemotcatic loop
d. NC, the number of iteration to be undertaken in a chemotactic loop (N_C>N_S)
e. N_{re} ,the maximum number of reproduction to be undertaken.
f. N_{ed} ,the maximum number of elimination and dispersal events to be imposed over bacteria.
g. p_{ed} ,the probability with which the elimination and dispersal will continue.
h. The location of each bacterium P(1-p,1-s,1) which is specified by random numbers within [-1,1].
i. The value of C (i), which is assumed to be constant in our case for all bacteria to simplify the design strategy.
j. The value of $d_{attract}$,$W_{attract}$,$h_{repelent}$ and $W_{repelent}$ .it is to be noted here that the value of $d_{attract}$ and $h_{repelent}$ must be same so that the penalty imposed on the cost function through "J_{CC}'' of (11) well be "0'' when all the bacteria will have same value , i.e. ,they have converged.

After initialization of all the above variables, keeping one variable changing and others fixed the value of "J'' proposed. Using eqn(9) the optimum cost function values are obtained by simulating the two-area interconnected power system without and with SSSC in the tie-line. Corresponding to the minimum cost, the magnitude of the changing variables is selected.

Table 1. BFO parameters

Sl.No	Parameters	Value
1	Number of Bacterium (s)	6
2	Swimming length (N_s)	3
3	Number of iteration in a Chemotactic loop (N_c)	10
4	Number of reproduction (N_{re})	15
5	Number of elimination and dispersal event (N_{ed})	2
6	Probability with which the elimination and dispersal(P_{ed})	0.25
7	Number of Parameters(P)	2
8	$W_{attract}$	0.04
9	$d_{attract}$	0.01
10	$h_{repellent}$	0.01
11	$\mathbf{W_{repelent}}$	10

Step - 2 Iterative algorithms for optimization:

This section models the bacterial population chemotaxis is Swarming, reproduction, elimination, and dispersal (initially, j=k=l=0).for the algorithm updating θ^i automatically results in updating of `P'.
(1) Elimination –dispersal loop: $l = l + 1$
(2) Reproduction loop: $k = k + 1$
(3) Chemotaxis loop: $j = j + 1$
(a) For i=1,2,…..S, calculate cost for each bacterium i as follows.

- Compute value of cost $J(i, j, k, j)$, Let

$$J_{sw}(i, j, k, l) = J(i, j, k, l) + J_{cc}(\theta^i(j, k, l), P(j, k, l))$$

[ie, add on the cell to cell attractant effect obtain through (12) for swarming behavior to the cost value obtained through(9)].

- Let $J_{last} = J_{sw}(i, j, k, l)$ to save this value since we may find a better cost via a run
- End of For loop.

(b) for i=1,2….S take the tumbling / swimming decision

- Tumble: generate a random vector $\Delta(i) \in \Re^p$ with each element $\Delta_m(i) m = 1,2,.....p$, a random number on [-1,1].

- Move let

$$\theta^i(j+1,\ k,\ l) = \theta^i(j,\ k,\ l) + C(i)\frac{\Delta(i)}{\sqrt{\Delta^T(i)\ \Delta(i)}} \quad (12)$$

Fixed step size in the direction of tumble for bacterium `i' is considered

- Compute $J(i, j+1, k, l)$ and then let

$$J_{sw}(i, j+1, k, l) = J(i, j+1, k, l) + J_{cc}(\theta^i(j+1, k, l), P(j+1, k, l)) \quad (13)$$

- Swim:
  (i) Let m=0 ;(counter for swim length)
  (ii) While m<Ns (have not climbed down too long)

- Let m=m+1
  If $J_{sw}(i, j+1, k, l) < J_{last}$ (if doing better), let
  $J_{last} = J_{sw}(i, j+1, k, l)$ and in eqn(12)
  And use this $\theta^i(j+1, k, l)$ to compute the new $J(i, j+1, k, l)$.

Else let m=N_s. This the end of while statement

c) Next bacterium (i+1) is selected if i ≠S (ie go to b) to process the next bacterium

4) If j< N_c, go to step 3 .In this case, chemotaxis is continued since the life of the bacteria in not over

5) Reproduction.

a) for the given k and *l* for each i=1,2……S let $J^i_{health} = \min_{j\in\{1...Nc\}}\{J_{sw}(i, j, k, l)\}$ be the health of the bacterium i (a measure of how many nutrients it)got over its life time and how successful if was at avoiding noxious substance).Sort bacteria in order of ascending cost J_{health} (higher cost means lower health).

b) the S_r =S/2 bacteria with highest J_{health} valves die and other S_r bacteria with the best value split [and the copies that are placed at the same location as their parent

6) it k<$N_{re,}$ go to 2; in this case ,as the number of specified reproduction steps have not been reached ,so we start the next generation in the chemotactic loop is to be started

7) Elimination –dispersal: for i = 1,2… S with probability Ped, eliminates and disperses each bacterium [this keeps the number of bacteria in the population constant] to a random location on the optimization domain

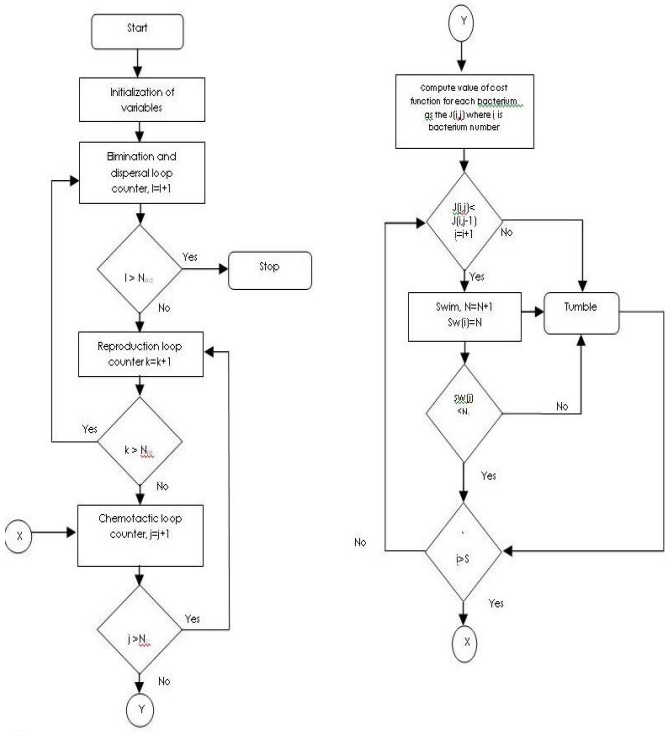

Figure 4. Flow Chart for bacterial foraging algorithm

5. Simulation Results and Observations

The optimal gain of Integral controllers (KI1, KI2), SSSC based damping controller gain (K) and time constants (T_1,T_2.T_3,T_4) of SSSC unit are determined on the basis of BFO technique. These controllers are implemented in a interconnected two-area power system without and with SSSC units for 1 % step load disturbance in area 1. The integral gain values, cost function values, settling time and peak over/under shoot for the frequency deviations in each area and tie-line power deviation for interconnected two-area power system without and with SSSC units are tabulated in table2. The optimal gain and time constant of SSSC based damping controller are found as K = 0.1; T_1 = 0.2651; T_2 = 0.2011; T_3 = 0.6851; T_4 = 0.2258; The output responses of the two-area interconnected system have been shown in fig 5-7. From fig 5-6, it is evident that the dynamic responses have Improved significantly with the use of SSSC units. Fig7 shows the generation responses considering apf11=apf12=0.5 and apf21= apf22 = 0.5with out and with SSSC units having Δf_1 as the control logic signals

From the tabulation, it can be found that the controller designed for two area thermal reheat power system with SSSC have not only reduces the cost function but also ensure better stability, moreover possesses less over/ under shoot and faster settling time when compared with the controller designed for the two area-two unit thermal reheat power system without SSSC. As the SSSC units, suppresses the peak frequency deviations of both areas, governor system continue to eliminate the steady state error of frequency deviations as expected.

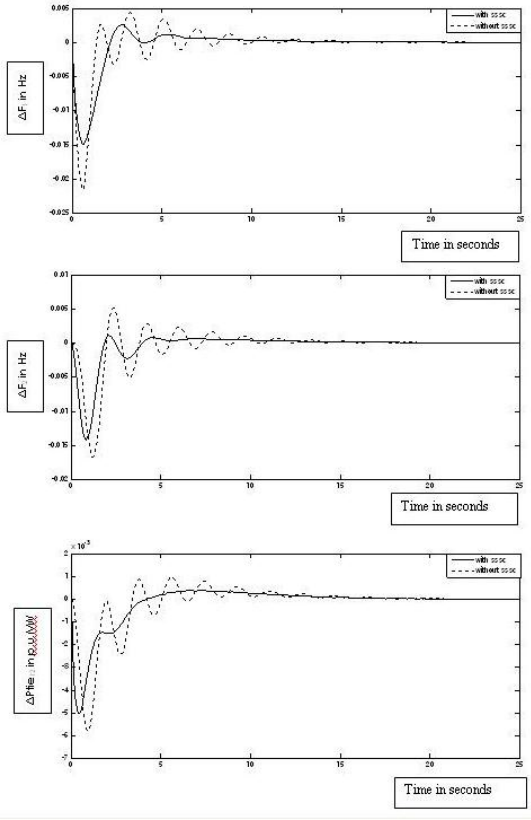

Figure 5. Dynamic responses of the frequency deviations and tie line power deviation considering a step load disturbance of 0.01p.u in area1.

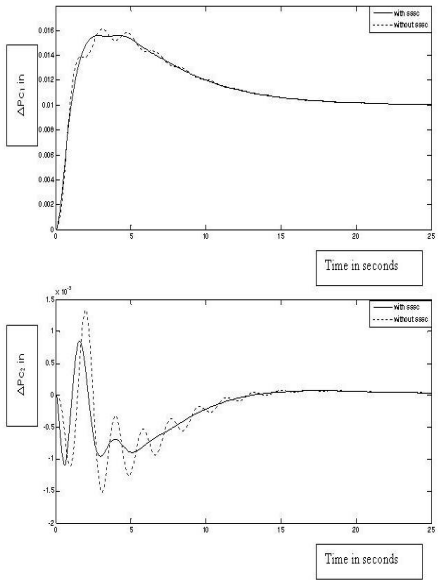

Figure 6. Dynamic responses of the Control input deviations considering a step load disturbance of 0.01p.u in area1.

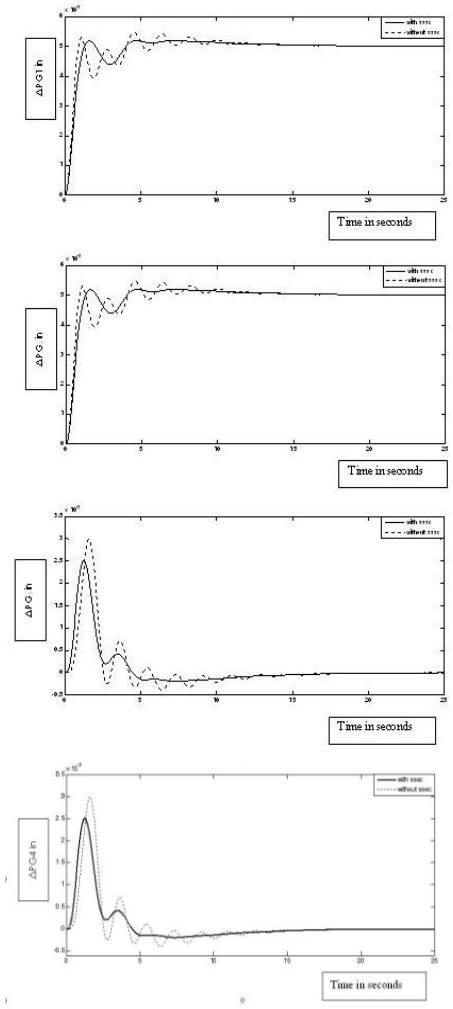

Figure 7. Dynamic responses of the required additional mechanical power generation for step load disturbance of 0.01p.u in area1

Table 2: Comparison of the system performance for the two case studies

Two area two unit interconnected thermal reheat power system under 0.01 p.u.MW step load disturbance in area 1	Feedback gain(K_i)	Cost function value [J]	Setting time (τ_s) in seconds			Peak over / under shoot		
			ΔF_1	ΔF_2	ΔP_{tie}	ΔF_1 in Hz	ΔF_2 in Hz	ΔP_{tie} p.u MW
Case:1 without SSSC unit	1.0848	0.5879	16.37	15.19	16.48	0.02157	0.01662	0.00578
Case:2 with SSSC unit	1.1656	0.3790	9.643	9.541	12.96	0.01488	0.01454	0.004943

Conclusion

In this paper, the responses of a two area interconnected, thermal reheat power system without and with SSSC units have been studied. Integral gain setting have been optimized by bacterial forging optimization technique. Small rating SSSC units are connected is series with tie-line of the two area interconnected power system and responses shows that they are capable of consuming the oscillations in area frequency deviations and tie-line power deviations of the power system. Further SSSC units reduce the over/under shoot and settling time of the output responses. Hence, it may be concluded that SSSC units are efficient and effective for improving the dynamic performance of load frequency control of inter connected power system than that of the system without SSSC unit.

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Nomenclature

f	Area frequency in Hz
J_i	Cost function of area i
k_r	Reheat coefficient of the steam turbine
k_I	Optimum Integral feedback gain
N	Number of interconnected areas
P_{ei}	The total power exchange of area i in p.u.MW / Hz
P_{Di}	Area real power load in p.u.MW
P_G	Mechanical (turbine) power output in p.u.MW
R	Steady state regulation of the governor in Hz / p.u.MW
s	Laplace frequency variable
T_p	Area time constant in seconds
T_g	Time constant of the governing mechanism in seconds
T_r	Reheat time constant of the steam turbine in seconds
T_t	Time constant of the steam turbine in seconds
X_E	Governor valve position in p.u.MW
B_i	Frequency bias constant in p.u.MW / Hz

Δ Incremental change of a variable

T_{sssc} Time constant of SSSC in seconds

Superscript

T Transpose of a matrix

Subscripts i, j Area indices $(i, j = 1,2,..N)$

References

[1] Shayeghi.H, Shayanfar.H.A, Jalili.A, "Load frequency Control Strategies: A state-of-the-art survey for the researcher", Energy Conservation and Management, Vol 50(2), pp.344-353, 2009

[2] Ibraheem I, Kumar P and Kothari DP, "Recent philosophies of automatic generation control strategies in power systems" IEEE Transactions on power system, Vol 20(1), pp. 346-357, 2005

[3] Malik OP, Ashok Kumar, Hope GS, "A load-frequency control algorithm based on a generalized approach" IEEE Transactions on Power Systems, Vol3(2), pp. 375-382, 1988

[4] Hiyama T., "Design of decentralized load frequency regulators for interconnected power system", IEE Proceedings, Vol1, pp.17-23, 1982

[5] Chidambaram IA, Velusami S, "Design of decentralized biased controllers for load-frequency control of interconnected power systems", International Journal of Electric Power Components and Systems, Vol 33(12), pp.1313-1331, 2005

[6] Gyugyi L, Schauder C, Sen K, "Static synchronous series compensator: a solid –state approach to the series compensation of transmission lines", IEEE Transaction on Power Delivery, Vol 12(1), pp. 406-417, 1997

[7] Issarachai Ngamroo, "A Stabilization of Frequency Oscillations in an interconnected Power System Using Static Synchronous Series Compensator" Thammasat Int J.Sc.Tech, Vol 6, No.1, pp.52-60, 2001

[8] Ngamroo. I, Kongprawechnon. W, "A robust controller design of SSSC for stabilization of frequency oscillations in inter connected power system", International Journal of Electric Power System Research,, Vol 67, pp.161-176, 2003

[9] Liang CH, Chung C.Y, Wong K.P, Duan X.Z,. Tse C.T, "Study of Differential Evolution for Optimal Reactive Power Dispatch", IET, Generation Transmission and Distribution,Vol1, pp.253-260, 2007

[10] Shi Y and Eberhart R.C, "Parameter Selection in PSO", Proceedings of 7th Annual Conference on Evolutionary Computation, pp.591-601. 1998

[11] Dorigo M and Birattari M and Stutzle T, "Ant Colony optimization: artificial ants as a computational intelligence technique", IEEE Computational Intelligence Magazine, pp.28-39, 2007

[12] Djukanovic.m, Novicevic.M, Sobajic.D.J., Pao.Y.P., "Conceptual development of optimal load frequency control using artificial neural networks and fuzzy set theory", International Journal of Engineering Intelligent System in Electronic Engineering Communication, Vol.3,No.2, pp.95-108, 1995

[13] Ghoshal, "Application of GA/GA-SA based fuzzy automatic control of multi-area thermal generating system", Electric Power System Research, Vol.70, pp.115-127, 2004

[14] Abido.M.A., "Optimal design of power system stabilizers using particle swarm optimization", IEEE Transaction Energy conversion, Vol.17, No.3, pp.406-413, 2002

[15] Janardan Nanda, Mishra.S., Lalit Chandra Saikia "Maiden Application of Bacterial Foraging-Based optimization technique in multi-area Automatic Generation Control", IEEE Transaction .Power Systems, Vol.24, No.2, 2009

[16] Passino.K.M, "Biomimicry of bacterial foraging for distributed optimization and control", IEEE Control System Magazine, Vol.22, No.3, pp.52-67, 2002

[17] Mishra.S and Bhende.C.N, "Bacterial foraging technique-based optimized active power filter for Load Compensation", IEEE Transactions on Power Delivery, Vol.22, No.1, pp.457-465, 2007

[18] Elgerd.O.I., Electric Energy Systems Theory and Introduction, 2nd edition, New Delhi, India; Tata MC Graw-Hill, 1983

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Appendix

System Data [5, 7]

Rating of each area = 2000 MW, Base power = 2000 MVA, f^o = 60 Hz,$R_1 = R_2 = R_3 = R_4$ = 2.4 Hz / p.u.MW, $T_{g1} = T_{g2} = T_{g3} = T_{g4}$ = 0.08 sec, $T_{r1} = T_{r2} = T_{r1} = T_{r2}$ = 10 sec, $T_{t1} = T_{t2} = T_{t3} = T_{t4}$ = 0.3 sec, $K_{p1} = K_{p2}$ = 120Hz/p.u.MW, $T_{p1} = T_{p2}$ = 20 sec, $\beta_1 = \beta_2$ = 0.425 p.u.MW / Hz, $K_{r1} = K_{r2} = K_{r3} = K_{r4}$ = 0.5, $2\pi T_{12}$ = 0.545 p.u.MW / Hz, a_{12} = -1, ΔP_{D1} = 0.01 p.u.MW, T_{SSSC} = 0.03sec

Motion Heuristics Approach of Search an Optimal Path from Source to Destination in Robots

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Abstract—One of the most important problem in robotics is the task planning problem. A task is a job or an application or an operation that has to be done by the robot, whether it is a stationary robot or a mobile robot. The word planning means deciding on a course of action before acting. Before a robot does a particular task, how the task has to be done or performed in its workspace has to be planned. This is what is called as Robot Task Planning. A plan is a representation of a course of action for achieving the goal. How the problem has to be solved has to be planned properly. Robot task planning is also called as problem solving techniques and is one of the important topics of Artificial Intelligence. For example, when a problem is given to a human being to be solved ; first, he or she thinks about how to solve the problem, then devises a strategy or a plan how to tackle the problem. Then only he or she starts solving the problem. Hence, robot task planning is also called as robot problem solving techniques. Many of the items in the task planning are currently under active research in the fields of Artificial Intelligence, Image Processing and Robotics. Lot of research is going on in the robot problem solving techniques. In this paper, a optimal path planning algorithm using motion heuristics search problem is designed for a robot in a workspace full of obstacles which are polyhedral and consisting of only triangular objects.

Keywords **—** Robot, Uncertainty, Optical distortion, Precision, Workspace fixture, Camera.

I. INTRODUCTION

A typical robot problem solving consists of doing a household work ; say, opening a door and passing through various doors to a room to get a object. Here, it should take into consideration, the various types of obstacles that come in its way, also the front image of the scene has to be considered the most. Hence, robot vision plays a important role. In a typical formulation of a robot problem, we have a robot that is equipped with an array of various types of sensors and a set of primitive actions that it can perform in some easy way to understand the world [37]. Robot actions change one state or configuration of one world into another. For example, there are several labeled blocks lying on a table and are scattered [40]. A robot arm along with a camera system is also there [38]. The task is to pick up these blocks and place them in order [39]. In a majority of the other problems, a mobile robot with a vision system can be used to perform various tasks in a robot environment containing other objects such as to move objects from one place to another ; i.e., doing assembly operations avoiding all the collisions with the obstacles [1].

The paper is organized as follows. A brief introduction about the work was presented in the previous paragraphs. Section 2 gives the interpretation of the design of the obstacle collision free path. The mathematical interpretation is developed in the section 3 with its graphical design. Motion heuristics is dealt with in section 4. The section 5 shows the simulation results. The conclusions are presented in section 6 followed by the references.

II. INRERPRETATION OF THE DESIGN OF THE OBSTACLE COLLISION FREE PATH

One of the most important method of solving the gross motion planning problem is to go on searching all the available free paths in the work space of the robot [35]. The space in between the obstacles is referred to as the freeways along which the robot or the object can move. Translations are performed along the freeways and rotations are performed at the intersection / junctions of freeways [36]. This is an efficient method of obtaining an obstacle collision free path in the work space of the robot from source to the goal and is defined as the locus of all the points which are equidistant from two or more than two obstacle boundaries as shown in the Fig. 1 [37]. Once, the obstacle collision free paths are obtained from then source to the goal, then, the shortest path is found using graph theory techniques, search techniques and the motion heuristics [2].

There are certain advantages / disadvantages of this method of gross motion planning. They are

- It generates paths for the mobile part that stays well away from the obstacles ; since, the path is equidistant or midway between the obstacles and avoids collision with the obstacles [3].

- This method of planning the path using gross motion technique is, it is quite effective especially when the workspace of the robot is sparsely populated with obstacles [34].
- The path obtained is the shortest path [33].
- The path is a obstacle collision free path [31].
- The path is equidistant from the obstacles and there is no chance of collisions [30].

The method also works successfully when the workspace is cluttered with closely spaced obstacles, as a result of which the designed graph becomes more complex [4].

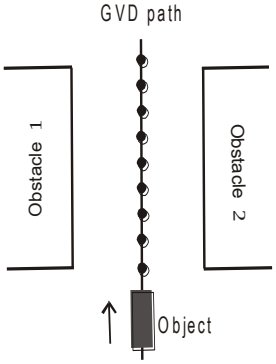

Fig. 1: Obstacle collision free path

III. MATHEMATICAL DEVELOPMENT & GRAPHICAL DESIGN OF THE PATH

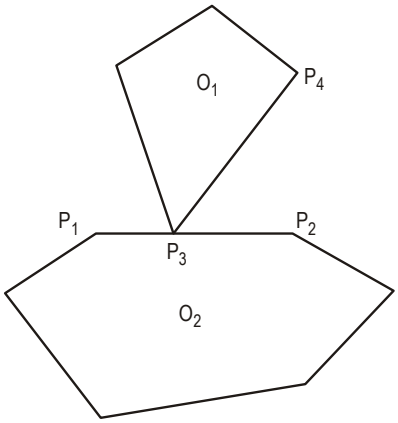

Fig. 2 : Interaction b/w a pair of edges of 2 obstacles O_1 and O_2

Here, in this section, we develop the mathematical interpretation of the obstacle collision free path. The robot work space consists of a number of obstacles. The parameters of any obstacles are the edges and the vertices [29]. So, while constructing the obstacle collision free path from the **S** to the **G**, many types of interactions occur [28]. Because, when we move from the source to the goal, we come across edges and vertices of many obstacles [5]. Here, we have considered only the interaction between a pair of edges of two obstacles as shown in the Fig. 2 [27], [1].

In this type of interaction between a pair of edges (interaction between an edge of one obstacle with an edge of another obstacle) as shown in the Fig. 2 [26]. How to construct the obstacle collision free path from S to the G when the obstacles are like this ? Consider two edges P_1P_2 and P_3P_4 of two obstacles O_1 and O_2 as shown in the Fig. 2. Here, P_3P_4 is an edge interacting with P_1P_2 at the point P_3 [6], [25].

P_1P_2 **;** P_3P_4 - Two edges of obstacles O_1 and O_2 meeting at the point P_3 [1]

R - Radius of the GVD cone.

λ - Be the distance parameter measured along the edge P_1P_2 measured from P_1.

l_0 - Distance from P_1 to P_3 along P_1P_2.

l_1 - Distance from P_1 to P_5 along P_1P_2.

l_2 - Length of P_3P_4.

d - Perpendicular distance from P_4 to P_1P_2.

The radius of the obstacle collision free path along P_1P_2 can be expressed by a piece-wise linear function of λ and is given by Eq. (1), where **sgn** denotes the signum function or the sign of the particular parameter l_2 and d, l_0 , l_1 and l_2 are as shown in the Fig. 3 [7], [24], [1].

$$R(\lambda) = \frac{(\lambda - l_0)\{l_0 - l_1 + \mathbf{sgn}(\lambda - l_0) l_2\}}{d} .$$

From this equation (1), we can come to a conclusion that [8]

If $(\lambda - l_0) > 0$; i.e., the point P is lying to the right of P_3 ; then, l_2 is positive , sgn ( ) is +.

If $(\lambda - l_0) < 0$; i.e., the point P is lying to the left of P_3 ; then, l_2 is negative , sgn ( ) is –

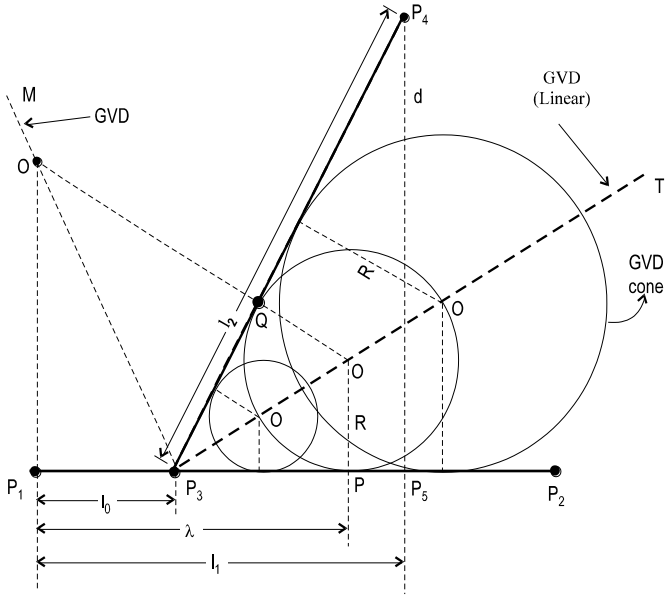

Fig. 3 Interaction between a pair of edges

First Method of Obtaining the obstacle collision free path :

Consider a point P any where on the line $P_1 P_2$ and which is at a radial distance of λ from P_1. Draw a $\perp^r$ line (dotted) from P [9]. Find the length of this line $\perp^r$ line (dotted) by using the Eq. (1) and mark the length (PO), where PO is the radius of the obstacle collision free path circle with O as the center. With O as center, draw a circle to pass through the points Q, P. Like this, go on taking different points (P's) on the line $P_1 P_2$ and which is at a radial distance of different λ's from P_1. Go on drawing $\perp^r$ lines from P. Go on finding the length of these $\perp^r$ lines (radii) using the formula, with their centers, go on drawing circles to touch the two edges. Go on joining all the centers of the obstacle collision free path circles. Thus, we get the obstacle collision free path from the source to the goal when the obstacles edges are straight lines [1].

Second Method of Obtaining the obstacle collision free path :

Bisect the angles $P_4 P_3 P_2$ and $P_4 P_3 P_1$ to get different point (O's). With these points as centers, draw the circles with the radii as the perpendicular distances [11]. Join the centers of all the circles, we get the obstacle collision free path, which is nothing but the angle bisector $P_3 T$ [12]. Similarly, we get another angle bisector $P_3 M$ along which the robot or the object would move, the path being perpendicular to the previous path [10], [1].

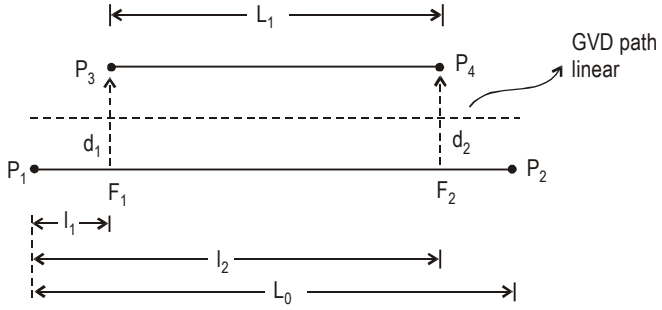

Fig. 4 : When the two edges $P_1 P_2$ and $P_3 P_4$ are parallel

Special Case of Interaction Between Two Edges :

When two edges $P_1 P_2$ and $P_3 P_4$ are parallel as shown in the Fig. 4, the obstacle collision free path is the mid-way path between the two edges of the obstacles [24]. Hence, to conclude, the interaction between a pair of edges is linear or a straight line [13]. If the robot or the object held by the tool / gripper moves along the obstacle collision free path, then definitely, there would be no collision of the object or the robot itself with the obstacles [14], [1].

IV. MOTION HEURISTICS

Motion heuristics is the method of searching an obstacle collision free path in the free work space of the robot from the source to the destination by making use of search techniques such as the graph theory (AND / OR graphs), chain coding techniques and the state space search techniques (best first search, breadth first search) used in Artificial Intelligence [23]. The search techniques used in AI to find the path from the source S to the goal G are called as motion heuristics or the robot problem solving techniques [15]. The word 'heuristic' means to search , what to search ? an obstacle collision free path to search [16].

V. PROBLEM SIMULATION & SIMULATION RESULTS

We consider a workspace cluttered with obstacles, especially triangular obstacles. These triangular obstacles are placed either on the table or on the floor, which is simulated on the computer as a 2D rectangular workspace [22]. Using the mouse or using a rectangular coordinates, we specify the source coordinates (x_1 , y_1) [17]. Similarly, using the mouse or using rectangular coordinates, we specify the destination coordinates (x_2 , y_2) [18]. A computer algorithm is written using the user-friendly language C++ to find the shortest path using the formula given in Eq. (1). The results of the simulation are shown in the Figs. 5 and 6 respectively [19]. The motion heuristics used in Artificial Intelligence is used to find the shortest path from the source to the goal [21]. Using this motion heuristics, a number of paths are available from the source to the goal, but it selects a shortest path which is the path shown in yellow color in the Fig. 6 [20], [1].

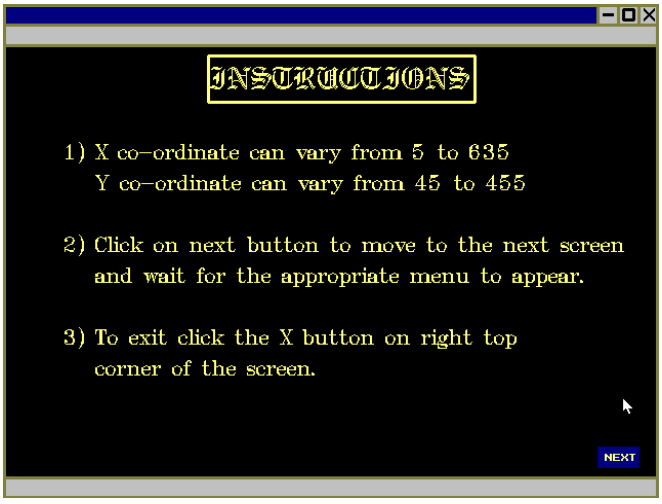

Fig. 5 : Instruction for entering the rectangular coordinates

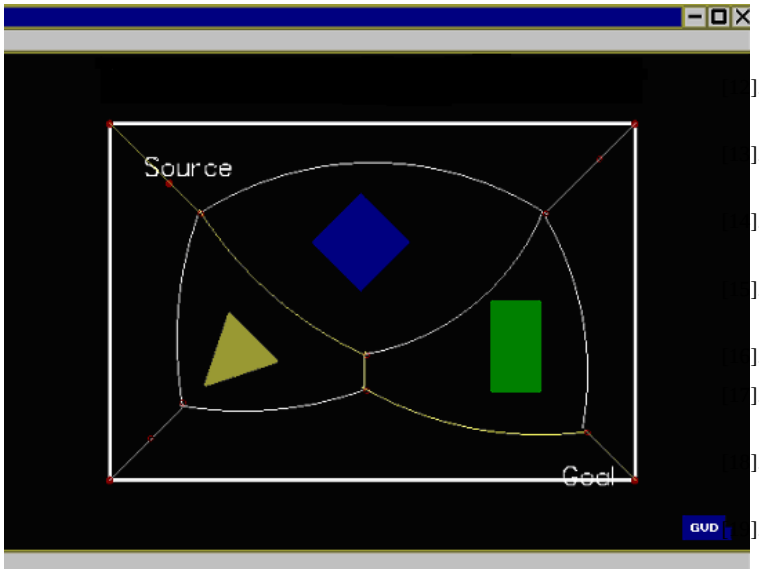

Fig. 6 : Graph showing all the available free paths from S to G

VI. CONCLUSION

A new method of finding an obstacle collision free path from the source to the goal when the workspace is cluttered with obstacles is developed using motion heuristics using an user friendly GUI developed in C++. This method is similar to the method of finding / searching a path by the humans. The method was also implemented on a real time system, say a robot and was successful. Thus, the Artificial Intelligence which uses motion heuristics (search methods) is used to find the obstacle collision free path.

REFERENCES

[1]. Robert, J.S., *Fundamentals of Robotics : Analysis and Control*, PHI, New Delhi., 1992.

[2]. Klafter, Thomas and Negin, *Robotic Engineering*, PHI, New Delhi, 1990.

[3]. Fu, Gonzalez and Lee, *Robotics : Control, Sensing, Vision and Intelligence*, McGraw Hill, Singapore, 1995.

[4]. Ranky, P. G., C. Y. Ho, *Robot Modeling, Control & Applications*, IFS Publishers, Springer, UK., 1998.

[5]. T.C.Manjunath, *Fundamentals of Robotics*, Nandu Publishers, 5th Revised Edition, Mumbai., 2005.

[6]. T.C.Manjunath, *Fast Track To Robotics*, Nandu Publishers, 3nd Edition, Mumbai, 2005.

[7]. Ranky, P. G., C. Y. Ho, *Robot Modeling, Control & Applications*, IFS Publishers, Springer, UK, 2005.

[8]. Groover, Weiss, Nagel and Odrey, *Industrial Robotics*, McGraw Hill, Singapore, 2000.

[9]. William Burns and Janet Evans, *Practical Robotics - Systems, Interfacing, Applications*, Reston Publishing Co., 2000.

[10]. Phillip Coiffette, *Robotics Series*, Volume I to VIII, Kogan Page, London, UK, 1995.

[11]. Luh, C.S.G., M.W. Walker, and R.P.C. Paul, *On-line computational scheme for mechanical manipulators*, Journal of Dynamic Systems, Measurement & Control, Vol. 102, pp. 69-76, 1998.

[12]. Mohsen Shahinpoor, *A Robotic Engineering Text Book*, Harper and Row Publishers, UK.

[13]. Janakiraman, *Robotics and Image Processing*, Tata McGraw Hill.

[14]. Richard A Paul, *Robotic Manipulators*, MIT press, Cambridge.

[15]. Fairhunt, *Computer Vision for Robotic Systems*, New Delhi.

[16]. Yoram Koren, *Robotics for Engineers*, McGraw Hill.

[17]. Bernard Hodges, *Industrial Robotics*, Jaico Publishing House, Mumbai, India.

[18]. Tsuneo Yoshikawa, *Foundations of Robotics : Analysis and Control*, PHI.

[19]. Dr. Jain and Dr. Aggarwal, *Robotics : Principles & Practice*, Khanna Publishers, Delhi.

[20]. Lorenzo and Siciliano, *Modeling and Control of Robotic Manipulators*, McGraw Hill.

[21]. Dr. Amitabha Bhattacharya, *Mechanotronics of Robotics Systems*, Calcutta, 1975.

[22]. S.R. Deb, *Industrial Robotics*, Tata McGraw Hill, New Delhi, India.

[23]. Edward Kafrissen and Mark Stephans, *Industrial Robots and Robotics*, Prentice Hall Inc., Virginia.

[24]. Rex Miller, *Fundamentals of Industrial Robots and Robotics*, PWS Kent Pub Co., Boston.

[25]. Doughlas R Malcom Jr., *Robotics ... An introduction*, Breton Publishing Co., Boston.

[26]. Wesseley E Synder, *Industrial Robots : Computer Interfacing and Control*, Prentice Hall.

[27]. Carl D Crane and Joseph Duffy, *Kinematic Analysis of Robot Manipulators*, Cambridge Press, UK.

[28]. C Y Ho and Jen Sriwattamathamma, *Robotic Kinematics ... Symbolic Automatic and Numeric Synthesis*, Alex Publishing Corp, New Jersey.

[29]. Francis N Nagy, *Engineering Foundations of Robotics*, Andreas Siegler, Prentice Hall.

[30]. William Burns and Janet Evans, *Practical Robotics - Systems, Interfacing, Applications*, Reston Publishing Co.

[31]. Robert H Hoekstra, *Robotics and Automated Systems*.

[32]. Lee C S G, *Robotics , Kinematics and Dynamics*.

[33]. Gonzalez and Woods, *Digital Image Processing*, Addison Wesseley.

[34]. Anil K Jain, *Digital Image Processing*, PHI.

[35]. Joseph Engelberger, *Robotics for Practice and for Engineers*, PHI, USA.

[36]. Yoshikawa T., *Analysis and Control of Robot Manipulators with Redundancy*, Proc. First Int. Symp. on Robotics Research, Cambridge, MIT Press (1984), pp. 735-748.

[37]. Whitney DE., *The Mathematics of Coordinated Control of Prosthetic Arms and Manipulators*, Trans. ASM J. Dynamic Systems, Measurements and Control, Vol. 122 (1972), pp. 303-309.

[38]. Whitney DE., *Resolved Motion Rate Control of Manipulators and Human Prostheses*, IEEE Trans. Syst. Man, Cybernetics, Vol. MMS-10, No. 2 (1969), pp. 47-53.

[39]. Lovass Nagy V, R.J. Schilling, *Control of Kinematically Redundant Robots Using* {1}*-inverses*, IEEE Trans. Syst. Man, Cybernetics, Vol. SMC-17, No. 4 (1987), pp. 644-649.

[40]. Lovass Nagy V., R J Miller and D L Powers, *An Introduction to the Application of the Simplest Matrix-Generalized Inverse in Systems Science*, IEEE Trans. Circuits and Systems, Vol. CAS-25, No. 9 (1978), pp. 776.

Analysis of Software Quality Models for Organizations

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Abstract
Software Quality model is a vital to obtained data so that actions can be taken to improve the performance. Such improvement can be measured quality, increased customer satisfaction and decreased cost of quality. Software metrics and quality models play a pivotal role in measurement of software quality. A number of well known qualities models are used to build quality software. Different researchers have proposed different software quality models to help measure the quality of software products. In our research, we are discussing the different software quality models and compare the software quality models with each other. Also a framework containing steps is proposed by authors. Some recommendations are also framed hereby in the following research paper.

Keywords
Software Quality Models, McCall model, Dromey's model, FURPS model, ISO 9126 model.

Objectives
To begin with there are some common objectives:-

- To analysis various software quality models w.r.t various attributes.
- The presence, or absence, of these attributes can be measured objectively.
- The degree to which each of these attributes is present reflects the overall quality of the software product.
- These attributes facilitate continuous improvement, allowing cause and effect analysis which maps to these attributes, or measure of the attribute.

1. INTRODUCTION
"Quality comprises all characteristics and significant features of a product or an activity which relate to the satisfying of given requirements". Software is critical in providing a competitive edge to many organizations, and is progressively becoming a key component of business systems, products and services. The quality of software products is now considered to be an essential element in business success. Furthermore, the quality of software product is very important and essential since for example in some sensitive systems – such as, real-time systems, control systems, etc. – the poor quality may lead to financial loss, mission failure, permanent injury or even loss of human life. There are several definitions for "software Quality" term, for examples, it is defined by the IEEE [1990] as the degree to which a system, component or process meets specified requirements and customer (user) needs (expectations). Pressman [2004] defines it as "conformance to explicitly stated functional and performance requirements, explicitly documented development standards, and implicit characteristics that are expected of all professionally developed software." The ISO, by contrast, defines "quality" in ISO 14598-1 [ISO, 1999] as "the totality of characteristics of an entity that bear on its ability to satisfy stated and implied needs," and Petrasch [1999] defines it as "the existence of characteristics of a product which can be assigned to requirements." There are a number of quality models in software engineering literature, each one of these quality models consists of a number of quality characteristics (or factors, as called in some models). These quality characteristics could be used to reflect the quality of the software product from the view of that characteristic. Selecting which one of the quality models to use is a real challenge. In this paper, we will discuss the contents of the following quality models:

1. McCall's Quality Model.
2. Boehm's Quality Model.
3. Dromey's Quality Model.
4. FURPS Quality Model.
5. ISO 9126 Quality Model.

In addition, we will focus on a comparison between these quality models, and find the key differences between them.

1.1 McCall software Quality Model

One of the more renown predecessors of today's quality models is the quality model presented by Jim McCall (also known as the General Electrics Model of 1977).McCall's quality model defines and identifies the quality of a software product through addressing three perspectives: (i) ***Product operation*** is the product's ability to be quickly understood, operated and capable of providing the results required

by the user. It covers correctness, reliability, efficiency, integrity and usability criteria. (ii) ***Product revision*** is the ability to undergo changes, including error correction and system adaptation. It covers maintainability, flexibility and testability criteria. (iii)***Product transition*** is the adaptability to new environments, distributed processing together with rapidly changing hardware.
It covers portability, reusability and interoperability criteria. Not all the software evolvability sub characteristics are explicitly addressed in this model. Analyzability is not explicitly included as one of the perceived aspects of quality. However, as the model is further detailed into a hierarchy of factors, criteria and metrics, some of the measurable properties and metrics are related to the achievement of analyzability, e.g. simplicity and modularity. Architectural integrity is not covered in the model. Moreover, none of the factors or quality criteria in the model is related to architectural integrity with respect to the understanding and coherence to the architectural decisions. This model is proposed for general application systems, and thus the domain-specific attributes are not explicitly addressed in the scope of the model.

1.2 ISO 9126 Software Quality Model

ISO 9126 is an international standard for the evolution of software. The standard is divided into four parts which address respectively the following subjects: Quality model, External metrics, internal metrics and quality in use metrics. ISO 9126 Part-1 is an extension of previous work done by McCall (1977), Boehm (1978), FURPS etc. ISO 9126 specifies and evaluates the quality of a software product in terms of internal and external software qualities and their connection to attributes. The model follows the factor-criteria-metric model and categorizes software quality attributes into six independent high-level quality characteristics: functionality, reliability, usability, efficiency, maintainability and portability. Each of these is broken down into secondary quality attributes, e.g. maintainability is refined into analyzability, changeability, stability, testability and compliance to standards, conventions or regulations. One may also argue if the enhancement-with-new features type of change is embedded within the types of modifications defined in the quality model, i.e. corrections, improvements or adaptations of the software to changes in environment, requirements and functional specifications.

1.3 Boehm's Software Quality Model

Boehm [1976, 1978] introduced his quality model to automatically and quantitatively evaluate the quality of software. This model attempts to qualitatively define the quality of software by a predefined set of attributes and metrics. Boehm's quality model represents a hierarchical structure of characteristics, each of which contributes to the total quality. The model begins with the software's general utility, i.e. the high level characteristics that represent basic high-level requirements of actual use. The general utility is refined into a set of factors and each factor is composed of several criteria which contribute to it in a structured manner. The factors include: (i) ***portability*****;** (ii) ***utility*** which is further refined into reliability, efficiency and human engineering; and (iii) ***maintainability*** which is further refined into testability, understandability and modifiability. Neither in the Boehm quality model is all the software evolvability sub characteristics explicitly addressed. Analyzability is partially addressed through the characteristic *understandability*, which describes that the purpose of the code is clear to the inspector. However, none of the factors or measurable properties describes the capability to analyze the impact at the software architecture level due to a change stimulus. Architectural integrity is not covered in the model.

1.4 Dromey's Software Quality Model

Dromey's proposes a working framework for evaluating Requirement determination, design and implementation phases. The framework consists of three models, i.e. *Requirement quality model*, *Design quality model* and *Implementation quality model*. The high-level product properties for the implementation quality model include: (i) *Correctness* evaluates if some basic principles are violated, with functionality and reliability as software quality attributes; (ii) *Internal* measures how well a component has been deployed according to its intended use, with maintainability, efficiency and reliability as software quality attributes; (iii) *Contextual* deals with the external influences on the use of a component, with software quality attributes in maintainability, reusability, portability and reliability; (iv) *Descriptive* measures the descriptiveness of a component, with software quality attributes in maintainability, reusability, portability and usability. In this model, characteristics with regard to process maturity and reusability are more explicit in comparison with the other quality models. However, not all the evolvability sub characteristics are explicitly addressed in this model. Analyzability is only partially covered within the *contextual* and *descriptive* product properties at individual

component level, though none of these product properties describes the capability to analyze the impact at the software architecture level due to a change stimulus. Architectural integrity is not fully addressed despite the *design quality model* takes into account explicitly the early stages (analysis and design) of the development process. The focus of the *design quality model* is that a design must accurately satisfy the requirements, and be *understandable*, *adaptable* in terms of supporting changes and developed using a mature process. However, it is not sufficient for capturing architectural design decisions. Extensibility is not addressed as an explicit characteristic to represent future growths. Testability is implicitly embedded in the *internal* product property. Domain-specific attributes are not addressed. Moreover, one disadvantage of the Dromey model is associated with reliability and maintainability, as it is not feasible to judge them before the software system is actually operational in the production area.

1.5 FURPS Software Quality Model

The characteristics that are taken into consideration in FURPS model [9] are:
(i) *Functionality* includes feature sets, capabilities and security;
(ii) *Usability* includes human factors, consistency in the user interface, online and context-sensitive help, wizards, user documentation, and training materials;
(iii) *Reliability* includes frequency and severity of failure, recoverability, predictability, accuracy, and mean time between failure (MTBF);
(iv) *Performance* prescribes conditions on functional requirements such as speed, efficiency, availability, accuracy, throughput, response time, recovery time, and resource usage;
(v)*Supportability* includes testability, extensibility, adaptability, maintainability, compatibility, Configurability, serviceability, installability, and localizability / internationalization. Architectural integrity is not covered in the model. None of the characteristics or sub characteristics in the model is related to architectural integrity with respect to the understanding and coherence to the architectural decisions. Moreover, one disadvantage of this model is that it fails to take account of the software portability. Domain-specific attributes are not addressed either in the model.

2. METHODOLOGY

This section discusses the chosen research methodology. A discussion of alternative research methods is included in Appendix at the last of conclusion.

I. The First method selected will use questionnaires to measures the impact of lifecycles on the factors that influence the outcomes of software project. The resultant data will be used to drive a model selection framework. Which will suggest how suggest how suitable a lifecycle model is for a given project. The framework recommendations will be examined and discussed using a set of case studies.

2 Questionnaires provide ‘a structured approach to gathering date’ and that ‘closed questions, providing a limited list of responses, ensures easy transcription for processing’. This makes closed questions suitable for gathering lifecycle impact data, while open questions can be used to solicit and would typically be gathered from free-text entry boxes.

3. In my experience, project lifecycle model choice is often made automatically and without consciously considering alternative models. This suggests volunteers may need time to gather their thoughts before answering model selection questions. Questionnaires are ideal in this situation, since there is no (realistic) set time limit for completing them.

3. ANALYSIS/COMPARISON

In this research paper, we have studied different types of software quality models like McCall, ISO 9126, Dromey’s etc. From the 17 characteristics, only one characteristic is common to all quality models, that is, the ‘reliability’. Also, there are only three characteristics (i.e. ‘efficiency’, ‘usability’ and ‘portability’) which are belonging to four quality models. Two characteristic is common only to three quality models, that is, the ‘functionality’ and ‘maintainability’ characteristics. Two characteristic belong to two quality models, that is, the ‘testability’ and ‘reusability’ characteristics. And, nine characteristics (i.e. ‘flexibility’, ‘correctness’, ‘integrity’ and ‘interoperability’ in McCall’s quality model; ‘human engineering’, ‘understandability’ and ‘modifiability’ in Boehm’s quality model; ‘performance’ and ‘supportability’ in FURPS quality model) are defined in only one quality model.
Furthermore, it can be noted that the ‘testability’, ‘interoperability’ and ‘understandability’ are used as factors/attributes/characteristics in some quality models. However, in ISO 9126-1, these factors/attributes/characteristics are defined as sub

characteristics. More specifically, the 'testability' is belonging to the 'maintainability' characteristic, the 'understandability' is belonging to the 'usability' characteristic, and the 'interoperability' is belonging to the 'functionality' characteristic. From our point of view, the ISO 9126-1 quality model is the most useful one since it has been built based on an international consensus and agreement from all the country members of the ISO organization.

1. There are some criteria's/ goals that support McCall model are: Correctness, Maintainability, Testability, Flexibility, Reliability, Usability, Interoperability
Reusability, Integrity, Efficiency, and Portability.

2. There are some criteria's/goals that support Boehm model are: Testability, Understandability, Efficiency Modifiability, Reliability, Portability, and Human Engineering.

3. There are some criteria's/goals that support ISO 9126 model are: Reliability, Maintainability, Portability, Usability, Functionality, and Efficiency.

4. There are some characteristics/attributes/goals that support FURPS model is: Reliability, Usability, Functionality Performance, and Supportability.

5. There are some characteristics/attributes/goals that support Dromey's model are: Maintainability, Reliability, Efficiency, Usability, Portability, Reusability, Functionality.

4. PROPOSAL

1. Define your organization's need and goals for software quality. If you don't know what your organizations need in terms of quality, you should not waste time on a software quality model. Answer the following questions:

   - What are your customer quality priorities?
   - Do you have processes in place to monitor customer preferences?
   - How do your customers feel about your products and services versus those of yours competitors?
   - Do you have special needs such as regulations or safety concerns?
   - Do you have special needs such as regulations or safety concerns?
   - Do you have contractual or legal requirements to use a particular model?

2. Identify which quality elements are most important to your business goals.
3. Choose a model. Based on elements you selected in step 2.
4. Develop details and examples to explain the software quality factors which are most important to your organization. These will help communicate priorities to your teams.
5. Build the quality factors and the quality model into your development and test methodologies.
6. If you have accomplished the steps above, you have organized a software development, test and quality process which systematically addresses the software quality elements which match your organization's strategic goals. But things change, so periodically recalibrate with your staff and customers to ensure agreement on goals and priorities.

4.1 Recommendations

Step1. Identify a small set of agreed-upon, high-level quality attributes, and then, in a top-down fashion decompose each attribute into a set of subordinate attributes.

Step2. Distinguish between internal and external metrics.

Step3. Identify type of users for each high-level quality attributes.

Step4. Put the pieces together; constructing the new models that implement ideas from international standards: ISO-9126, Drome, ISO.IEC TR 15504-2, and accordingly recognize appropriate Stakeholders for each set of attributes.

5. CONCLUSION

We have studied different types of software quality models in software engineering each of these quality models consists of number of characteristics. Selecting which one of the quality models to use is a real challenge. In this paper, we have discussed and compared the following quality models:

1. McCall's Quality Mode.
2. Boehm's Quality Model.
3. Dromey's Quality Model.
4. FURPS Quality Model.
5. ISO 9126 Quality Model

Based on the discussion of the five quality models and on the comparison between them, the following comments could be written:

1. In McCall's quality model, the quality is subjectively measured based on the judgment on the person(s) answering the questions ('yes' or 'no' questions).

2. Three of the characteristics are used in the ISO 9126-1 quality model as sub-characteristics from other characteristics.

3. The FURPS quality model is built and extended to be used in the IBM Rational Software Company. Therefore, it is a special-purpose quality model, that is, for the benefits of that company.

The ISO 9126-1 quality model is the most useful one since it has been build based on an international consensus and agreement from all the country members of the ISO organization.

6. Future Work

After studied of all these models we can build a new software quality model. During creating a new model the analysis step assisted us to benefit from existing general quality models and simultaneously avoiding repetition of such limitations.

7. ACKNOWLEDGMENTS

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REFERENCES

[1] Hoyer, R. W. and Hoyer, B. B. Y., "*What is quality*?" *Quality Progress*, no. 7, pp. 52-62, 2001.

[2]. Humphrey, Watts, "*The Software Quality Profile*", Software Engineering Institute. Carnegi-Mellon University. Nov. 2001. http://www.sei.cmu.edu/publications;/articles/qualtiy-profile/index.html

[3] Geoff, D., "*A model for Software Product Quality",* IEEE Transactions on Software Engineering, 21(2nd): 146-162.1995.

[4] B. W., Brown,et.al, "Characteristics of Software Quality. North Holland Publishing, Amsterdam, The Netherlands, 1978.

[5] M. Broy, F. Deissenboeck, and M. Pizka. Demystifying maintainability. In Proc. 4th Workshop on Software Quality (4-WoSQ), pages 21–26. ACM Press, 2006.

[6] R. G. Dromey, "*A model for software product quality*", IEEE Transactions on Software Engineering, 21(2):146–162, 1995.

[7] Breivold, H.P. et.al, '*Using Software Evolvability Model for Evolvability Analysis*', Mälardalen University, 2008.

[8] Forrest Shull, et.al, "*Contributions to Software Quality*", IEEE Software, vol. 23, no. 1, pp. 16-18, Jan/Feb, 2006.

[9] R. G. Dromey, "A *model for software product quality".* IEEE Transactions on Software Engineering, 21(2):146–162, 1995.

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Minimization of Number of Handoff Using Genetic Algorithm in Heterogeneous Wireless Networks

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***Abstract*:** A new vertical handoff decision algorithm is proposed to minimize the number of handoff in heterogeneous wireless networks, which comprise a number of wireless networks. In this paper we have proposed a multi criteria vertical handoff decision algorithm which will select the best available network with optimized parameter values (such as cost of network should be minimum) in the heterogeneous wireless network. The decision problem is formulated as multiple objective optimization problems and simulated using genetic algorithm. The simulation result shows that the number of handoff can be minimized if we take optimized network parameter values.

***Keywords*:** vertical handoff, genetic algorithm, no of handoff, heterogeneous network, neural network, parameter optimization

1. Introduction

The emergence and development of mobile devices continues to expand and reshape our living standards. In the recent years, advances in miniaturization, low-power circuit design, and development in radio access technologies and increase in user demand for high speed internet access are the main aspects leading to the deployment of a wide array of wireless and mobile networks. The varying wireless technologies are driving today's wireless networks to become heterogeneous and provide a variety of new applications (such as multimedia) that eases and smoothes the transition across multiple wireless network interfaces. Fourth generation wireless communication system is the promising solution for heterogeneous wireless networks. A heterogeneous (or hybrid) network can be defined as a network which comprises of two or more different access network technologies (VANET, WLAN, UMTS, CDMA, MANET) to provide ubiquitous coverage. The 4G wireless system has the potential to provide high data transfer rates, effective user control, seamless mobility.

Many internetworking mechanisms have been proposed [1]-[4] to combine different wireless technologies. Two main architectures (a) Tightly coupled (b) Loosely-coupled have been proposed for describing internetworking of heterogeneous networks. However, roaming across the heterogeneous networks creates many challenges such as mobility management and vertical handoff, resource management, location management, providing QoS , security and pricing etc. In this kind of environment, mobility management is the essential issue that supports the roaming of users from one network to another. One of the mobility management component called as handoff management, controls the change of the mobile terminal's point of attachment during active communication [5].

Handoffs are extremely important in heterogeneous network because of the cellular architecture employed to maximize spectrum utilization. Handoff is the process of changing the channel (frequency, time slot, spreading code etc.) associated with the current connection while a call is in progress. Handoff management issues [6] include mobility scenarios, decision parameters, decision strategies and procedures. Mobility scenarios can be classified into horizontal (between different cells of the same networks) and vertical (between different types of network) .In homogeneous networks, horizontal handoffs are typically required when the serving access router becomes unavailable due to mobile terminal's movement. In heterogeneous networks, the need for vertical handoff s can be initiated for convenience rather than connectivity reasons. The decision may depend on various groups of parameters such as network-related, terminal related, user-related and service related. The network-related parameters are mainly defined as bandwidth, latency, RSS, SIR (Signal to inference ratio), cost, security etc. The terminal related parameters are velocity, battery power, location information etc. User related deals with user profile and preferences, service capacities, QoS etc. A number of vertical handoff decision strategies [4]such as (1)traditional(2) function-based (3)user-centric(4)Multiple attribute decision (5)Fuzzy logic-based(6) neural networks-based and context-aware have been proposed in the literature .The handover procedures can be characterized as hard or soft handoff .The handoff can be hard when the mobile terminal is connected to only one point of attachment at a time whereas the handoff can be soft when the mobile terminal is connected to two point of attachment.

The process of vertical handoff can be divided into three steps, namely system discovery, handoff decision and handoff execution. During the system discovery, mobile terminal equipped with multiple interfaces have to determine which networks can be used and what services are available in each network. During the handoff decision phase, the mobile device determines which network it should connect to. During the handoff execution phase, connections are needed to be re-routed from the existing network to the new network in a seamless manner .This requirement refers to the Always Best connected (ABC) concept, which includes the authentication ,authorization , as well as the transfer of user's context information.

This paper presents the vertical handoff management and focuses mainly on the handoff decision problem. It is necessary to keep the decision phase in the global phase and to prove its contributions in the optimization of vertical handoff performance. For instance, the first choice can minimize the handoff latency, operation cost and avoid unnecessary handoffs. The second choice can satisfy network requirement such as maximizing network utilization. The third choice can satisfy user requirement such as providing active application with required degree of QoS. This process needs decision factors: decision criteria, policies, algorithms, control schemes. The decision criteria mentioned previously have to be evaluated and compared to detect and to trigger a vertical handoff. To handle [4] this problem many methodologies such as policy-enabled scheme, fuzzy logic and neural network concepts, advanced algorithms such as multiple attribute decision making, context-aware concept etc. have been explored.

The rest of the paper is organized as follows. We first describe the related works that has been done till date which helped us to propose the new approach. The next section describes the details of vertical handoff process and the heterogeneous wireless networking system model. At last the simulation results have been defined for the proposed approach, followed by the conclusion and future work.

1.1 Related work

The vertical handoff decision algorithms that are proposed in the current research literature can be divided into different categories. The first category is based on the traditional strategy of using the received signal strength (RSS) combined with other parameters. In [8], Ylianttila et al. show that the optimal value for the dwelling timer is dependent on the difference between the available data rates in both networks. Another category uses a cost function as a measurement of the benefit obtained by handing off to a particular access network. In [9], the authors propose a policy-enabled handoff across a heterogeneous network environment using a cost function defined by different parameters such as available bandwidth, power consumption, and service cost. The cost function is estimated for the available access networks and then used in the handoff decision of the mobile terminal (MT). Using a similar approach as in [8], a cost function – based vertical handoff decision algorithm for multi-services handoff was presented in [10]. The available network with the lowest cost function value becomes the handoff target. However, only the available bandwidth and the RSS of the available networks were considered in the handoff decision performance comparisons. The third category of handoff decision algorithm uses multiple criteria (attributes and/or objectives) for handover decision. An integrated network selection algorithm using two multiple attribute decision making (MADM) methods, analytical hierarchy process(AHP) and Grey relational analysis (GRA), is presented in [11] with a number of parameters. Multiplicative Exponent Weighting (MEW), Simple Additive Weighting (SAW), and Technique for Order Preference by Similarity to Ideal Solution (TOPOSIS) [12] algorithm allow a variety of attributes to be included for vertical handoff decision. Simulation results show that MEW, SAW and TOPSIS provide similar performance to all four traffic classes (conversational, streaming, interactive and background). GRA provides a slightly higher bandwidth and lower delay for interactive and background traffic classes. In [13], Nasser et al. propose a vertical handoff decision function that provides handoff decision when roaming across heterogeneous wireless networks.

The fourth category of vertical handoff decision algorithm uses computational intelligence techniques. In [14], an Artificial Neural Network (ANN) is used to control and manage handoffs across heterogeneous wireless networks. The proposed method is capable of distinguishing the best existing wireless network that matches predefined user preferences set on a mobile device when performing a vertical handoff. A fuzzy logic inference system has been proposed [15] to process a multi-criteria vertical handoff decision metrics for integration and interoperation of heterogeneous networks. In [16], two vertical handoff (VHO) decision making schemes has been proposed based on fuzzy logic and neural networks. In [17], a mobility management was proposed in a packet-oriented multi-segment using Mobile IP and fuzzy logic concepts. Fuzzy logic systems and neural network classifiers are good candidates for pattern classifiers due to their non-linearity and generalization capabilities. The fifth category is based on the knowledge of the context information of the mobile terminal and the networks in order to take intelligent and better decisions [18]. In [19], the authors present a framework with an analytical context categorization and a detailed handover decision algorithm.

2. Vertical handover decision system configuration

The aim of our approach is to design an intelligent system that has the ability to select the best available wireless network by considering user preferences, device capabilities and wireless features for handling vertical handoff in heterogeneous wireless environment. Here, we consider that the mobile node is moving in an overlapping area covered by a group of wireless networks providing small and large coverage area and managed by different service providers. The mobile node may run a VOIP application and video conference that requires an appropriate QoS level. Networks are divided into three categories: Home Network (HN), which is the network in which the mobile node has initiates its connection, the target networks (TNs) which are the networks to which mobile nodes intend to roam into, and the selected network (SN), which is the best network chosen by the mobile node using the intelligent scheme described in this paper.

A mobile node can be existing at a given time in the coverage area of an UMTS alone. But due to mobility ,it can move into the regions covered by more than one access networks, i.e. simultaneously within the coverage areas of , say , an UMTS BS and an 802.11 AP. Multiple 802.11 WLAN coverage areas are usually contained within an UMTS coverage area. So, at any given time , the choice of an appropriate attachment point(BS or AP) for each MN

needs to be made , and with vertical handoff capability the service continuity and QoS experience of the MN can be significantly enhanced.

Generally, the performance parameters of vertical handoff algorithms are (a) handover delay (b) number of handovers (c) handover failure probability (d) throughput. In our model, we have taken into consideration the following network parameters for vertical handoff decision function (i)bandwidth(B) (ii)latency(L) (iii)signal-to-noise ratio(SNR) (iv)throughput(TH) (v)cost(C) (vi)power consumption(P) and the network with minimum latency, cost, SNR and power consumption and maximum throughput will be selected, so that an appropriate QoS level can be maintained and the number of handoff can be minimized for all the networks.

This proposed technique consists of two parts. The first part defines a neural network approach to select a suitable access network once the handoff initiation algorithm indicates the need to handoff from the home network to a target network. The network selection decision process is formulated as a Multiple Attribute Decision Making (MADM) problem that deals with the evaluation of a set of alternative access networks using a multiple attribute access network selection (MANSF) defined on a set of attributes (parameters). The MANSF is an objective function that measures the efficiency in utilizing radio resources by handing off to a particular network. The MANSF is triggered when any of the following events occur: (a) a new service request is made(b) a user changes his/her preferences (c) the MN detects the availability of a new network (d) there is severe signal degradation or complete signal loss of the current radio link. The network quality Qi, which provides a measure of the appropriateness of certain network i is measured via the function:

$$Q_i = f\{Bi, 1/Li, 1/SNRi, THi, 1/Ci, 1/Pi\} \quad (1)$$

In order to allow for different circumstances, it is necessary to weigh each factor relative to the magnitude it endows upon for vertical handoff decision. Therefore, a different weight is introduced as follows:

$$Q\,i = f\{wb * Bi, wl * 1/Li, wsn * 1/SNi, wth * THi, wc * 1/Ci, wp * 1/Pi\} \quad (2)$$

Where wb, wl, wsn wth, wc,wp are the weights for each network and device parameters respectively. The values of these weights are fractions between 0 and 1. The sum of all these weights are equal to 1.Each weight is proportional to the significance of a parameter to the vertical handoff decision. The larger the weight of a specific factor, the more important that factor is to the user and vice versa. The optimum wireless access network must satisfy:

Maximize Qi (p),

Where Qi(p) is MANSF calculated for each network i, and p is the input vector parameters. Due to the fact that each of the preferences chosen by the user has an associated unit that is different from the other (cost is measured in Rs, power consumption is measured in watt etc.), it is necessary to find a way for equation (2) to generate an optimized output using associated weights. The network selection algorithm has been implemented using Linear Vector Quantization (Mat lab 6 help) neural network model. The performance of the algorithm has been measured by using the number of handoff parameter for all networks. This approach can be considered as the non-optimized technique for vertical handoff decision. The second part defines an approach to minimize the total number handoffs in the complete heterogeneous wireless network environment. The selection algorithm select that network for the handoff where the bandwidth, power consumption, signal-to noise ratio, handoff latency, operation cost is minimum and throughput is maximum. Basically the problem has been considered as an optimization problem that can be represented as

Minimize $\sum Xi$,

Where Xi, is the number of handoff evaluated for the network i. This problem has been implemented using genetic algorithm and simulated using genetic algorithm pattern search tool box (Mat lab 6).

3. Simulation

In this section, we provide the evaluation parameters used to analyze the performance of the proposed schemes. In our work we consider that mobile nodes are moving uniformly in an area covered by a set of networks managed by six network service providers (NSPi, i=1..6). The simulation scenario consists of the following access networks GSM, CDMA, WiMax, for macro cell, WiFi for micro cell, Bluetooth and LAN for pico cell. A mobile device is busy in downloading some audio and video files from the internet while moving in the environment. Then, the first proposed algorithm will select a target network from the list of available networks by taking the non-optimized parameter values of all the networks in the integrated heterogeneous environment. As defined previously, the scheme is simulated using a neural network approach.

Data used in this paper was customized in a way to suit the purpose of this method. Bandwidth values are taken in the ranges[14…10000]k bits/s, latency values are provided by the networks is in the range of[.3..600]ms, cost values are in the range of [227..2700]Rs, SN ratio values are in the range-12.5 100],power consumption values are in the range[2..340]db and throughput values are in the range[22..144000] kbps .We assume that the user is running a VoIP application ,which needs a stable amount of latency and consumption of power. There were 120 samples each containing six feature of the wireless networks and a seventh feature representing the type of network. The simulation has been carried out using about 120 samples. The simulation result that the total number of handoff is 88 independent of percentage of training data and testing data without optimizing the network parameters.

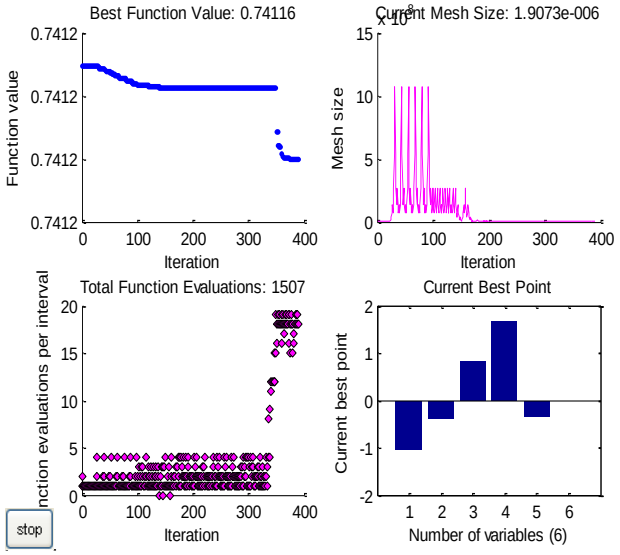

Figure 1

In the second scheme we have taken the optimized value of different parameters so as to minimize the total number of handoff for all the networks .As described previously ,we have simulated the scheme using genetic algorithm (psearchtool of Matlab 6).The simulation result shows that the number of handoff is less if all the parameters are optimized. The problem is represented as a minimization problem with the constraints such as that network will be selected where the bandwidth requirement ,power consumption ,handoff latency, signal to noise ratio and cost is minimum and the throughput is maximum. The figure 1(current best point graph) shows the total number of handoff is less (about 50) in comparison to non-optimized parameter values. The positive values shows the number of good handoff and the negative values shows the number of bad handoff. The best-function value and the mesh size graph shows that the function value is decreasing means the number of handoff is decreasing if the optimize parameter values are taken during handoff decision.

4. Conclusion and future work

4G in its evolutionary and revolutionary context does not allow an exact vision of the future. However, if past evolutionary developments are an indication of the future, there is a need to promote technological adaptability and interoperability for the next generation of wireless communications. This paper presents the design and performance issues for achieving an adaptable vertical handoff in heterogeneous 4G environment. Traditional handoff protocols are not sufficient to deal with the goal of seamless mobility with context-aware services. In this paper we have presented a context-aware vertical handoff scheme for 4G heterogeneous wireless communication environment. It uses a wide range of context information about networks, users, user devices and user applications and provides adaptations to a variety of context changes which are applicable to static and mobile users. The proposed handoff approach can meet the following requirements of handoff in heterogeneous wireless network(a)handoff is done fast and its delay is as less as possible (b)Number of handoff is minimized, which avoids degradation in signal quality and additional loads of the network(c)Handoff procedure is reliable and successful (d)Handoff algorithm is simple and has less computational complexity etc. This handoff scheme can be extended further to optimize the QoS for different types of multimedia applications in heterogeneous wireless environment.

References

1. 3GPP TR 23.234 v7.1.0, “3GPP System to WLAN Internetworking;System Description (Release 7),”March 2006, http://www.3gpp.org/specs/specs.html
2. A.K. Salkintzis,”Internetworking Techniques and Architectures for WLAN/3G Integration Toward 4G Mobile Data Networks,” IEEE Wireless Communications Magazine,vol.11,no. 3,pp.50-61,June 2004
3. M. Buddhikot,G. Chandramenon, S. Han, Y. W. Lee, S. Miller, and L. Salgarellim , “Integration of 802.11 and Third –Generation Wireless Data Networks,”IEEE Infocom ’03, vol. 1,pp. 503-512, San Francisco, USA, March 2003.
4. V. Verma, S. Ramesh, K. Wong, and J. Friedhoffer,” Mobility Management in Integrated UMTS/WLAN Networks”, IEEE ICC’03, vol. 2 pp. 1048-1053, Ottawa, Cannada, May 2003.
5. I. Akyildiz , J. Xie, S. Mohanty, A survey of Mobility Management in Next-Generation all-IP-based Wireless Systems, IEEE Wireless Communication 11(4) (2004) 16-28.
6. M. Kassar, Brigitte Kervella, Guy Pujolle,” An Overview of Vertical Handover Decision Strategies in Heterogeneous Wireless network,” ScienceDirect, Elsevier, Jan 2008.
7. “IEEE 802.21 Standard and Metropolitan area Networks: Media Independent Handover Services,” Draft P802.21/D00.05,January 2006.
8. M. Ylianttila et al., “Optimization scheme for mobile Users Performing Vertical Handoffs between IEEE 802.11 and GPRS/EDGE Networks”, Proc. Of IEEE GLOBECOMM’01, San Antinio, Texas, USA , Nov 2001.
9. H. Wang et al., “Policy –enabled Handoffs across Heterogeneous Wireless Networks”, Proc. Of Mobile Comp. Sys. and Apps., New Orleans, LA , Feb 1999
10. F. Zhu and J. McNair, “Vertical handoffs in fourth –Generation Multinetwork Environments”, IEEE Wireless Communications,June 2004.
11. Q. Song and A. Jamalipur ,” Network Selection in an integrated Wireless LAN and UMTS Environment using Mathematical Modeling and Computing Techniques “, IEEE Wireless Communications, June 2005.
12. M.Stoyanova & P.Mahonen, ’Algorithmic approaches for vertical handoff in Heterogeneous wireless environment’, IEEE WCNC 2007.
13. N. Nasser , A. Hasswa, and H. Hassanein ,” handoffs in Fourth Generation Heterogeneous networks”, IEEE Communications Magazine , October 2006
14. N. Nasser ,Bader Al.Manturi & H. Hassanein, ’A performance comparison of cross-based scheduling algorithms in future UMTS access’, Proc. of the IEEE IPCCC, Phoenix, Afizona, April 2005.
15. F. Zhu and J. McNair, ’Optimizations for vertical handoff decision algorithms’, IEEE WCNC, March2005.

16. M.Ylianttila,J.Makela & P.Mohenen, 'Optimization scheme for mobile users performing vertical handoffs between IEEE 802.11 and GPRS/EDGE network', IEEE PIMRC, vol1, sept. 2002.
17. P.M.L. Chan et al., "Mobility Management Incorporating Fuzzy Logic for a Heterogeneous IP Environment ", IEEE Communications Magazine, December 2001.
18. Q. Wei., K. Farkas, C. Prehofer, P. Mendes, B. Plattner, Context-aware Handover Using Active Network Technology, Computer Network 50 (15) 2006 2855-2872.
19. S. Balsubramaniam, J. Indulska, Vertical handover Supporting Pervasive Computing in future Wireless Networks, Computer Communications 27 (8) (2004) 708-719.
20. B.Kaur, urvashi Mittal,"Optimization of TCP using Genetic Algorithm,"Advances in Comutational Sciences and Technology,ISSN 0973-6107, Vol. 3 ,No. 2,(2010) pp119-125.
21. QiangMeng, Genetic Algorithm:Basics
22. HartMut Pohleim, "Genetic and Evolutionary Algorithms: Principles, Methods and Algorithms" http://www.geatbx.comldocu/algindex.html".

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Enhanced Developments in Wireless Mobile Networks (4G Technologies)

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Abstract **-** The latest mobile technology must have new features. With the advent of the Internet, the most-wanted feature is better, faster access to information. Cellular subscribers pay extra on top of their basic bills for such features as instant messaging, stock quotes, and even Internet access right on their phones. To support such a powerful system, we need pervasive, high-speed wireless connectivity. A number of technologies currently exist to provide users with high-speed digital wireless connectivity; Bluetooth and 802.11 are examples. The introduction of 4G has widened the scope of mobile communication. Now mobile is not only a device used for talking but it's more or less a portable computer that can serve different purposes. 4G offers higher data rates with seamless roaming. The mobile user can communicate without any disturbance while switching his coverage network. 4G is still passing through research and therefore there are some problems that need to be fixed in order to benefit the users from it fully. In this report we discuss various challenges 4G is facing and solutions to those problems are discussed. We propose our own way of improving QoS in 4G by using combination of mobility protocol SMIP and SIP. We propose that by using such scheme we can achieve better QoS during the process of handover.

Key Words

4G resp 3G: 4th (resp. 3rd) Generation
CDMA: Code Division Multiple Access
TDMA: Time Division Multiple Access
MIMO: Multiple Input Multiple Output
QoS: Quality of Service
OFDMA Orthogonal Frequency Division Multiple Access.
WiBro: Wireless Broadband
MANET: Mobile Ad-Hoc Network

1. Introduction

Mobile communication means communicating while on move. Mobile communication itself has seen various developmental stages such as first generation (1G), second generation (2G), third generation (3G) and fourth generation (4G). The Brief description of the generations of mobile communication is given in the bellow.

1G First generation of network came into use for the first time in July 1978 in USA.1G consisted of distributed transceivers that helped in communicating with mobile phone. The structure of the mobile phone was analogue and it could only be used for voice traffic. For the transmission of signals frequency modulation was in use. There was one 25MHZ frequency band allocation from cell base station to the mobile phone and another 25MHZ frequency band allocation for the signal from phone to the base station. In order to accommodate more users to the network each channel was separated from the other by a spacing of 30KHZ, but it was not effective enough in terms of the available spectrum.

2G The first 2G system was introduced in Finland in 1991, by Radiolinja (now part of Elisa Oyj). In 2G the shift was made to fully digital encrypted communication rather than analogue in 1G. 2G solved the problem of higher number of active customers in the network. Now more users could use the service simultaneously. 2G also introduced the additional data transfer through mobile rather than only voice data as in 1G. For example SMS text messages. As an example of successful 2G system we can study GSM, it was developed in 1980s and is currently under control of ETSI.

3G

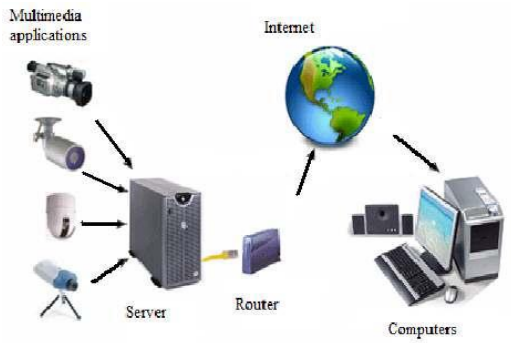

Figure 1 3G applications

3G can provider higher data rates both in mobile and in fixed environments. It gives up to 2Mbps in stationary and about 384 Kbps in mobile environments. 3G has encouraged the video streaming and IP telephony to develop further and provide cost effective services to mobile users. 3G is the ITU standard to represent third generation mobile telephone system under the scope international mobile telecommunication program (IMT2000). 3G can implement various network technologies such as UMTS, GSM, CDMA, WCDMA, CDMA200, TDMA and EDGE.

4G Fourth generation (4G) also called Next Generation Network (NGN) offers one platform for different wireless networks. These networks are connected through one IP core. 4G integrates the existing heterogeneous wireless technologies avoiding the need of new uniform standard for different wireless systems like World Wide Interoperability for Microwave Access (WiMAX), Universal Mobile Telecommunications System (UMTS), wireless local area network (WLAN) and General Packet Radio Service (GPRS). 4G networks will increase the data rates incredibly, by providing 100Mbps to 1Gbps in stationary and mobile environment respectively. In 4G the latency will be decreased considerably, because of all IP environments. 4G can be considered as a global network where users can find voice, data and video streaming at anytime and anywhere around the globe. In 4G the integration of network and its applications is seamless therefore there is no risk of delay. While implementing 4G the cost issue needs to be taken into consideration so that users can benefit from this technological development fully.

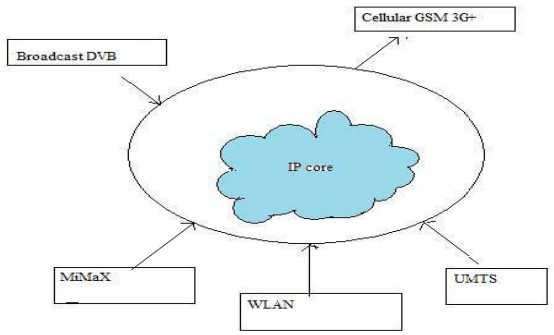

Figure 2 4G network

Present technology especially in areas of memory, bandwidth, and power, as well as new technological solution that should be available in near future are investigated in this paper. This paper should be able to present a picture of the physical constraints of MANET at present and also suggests some areas where previously considered as limitations may no longer exist, or will vanish in the near future. Hopefully this will allow other researchers to set reasonable constraints and physical boundaries for future research, tests, and simulations in MANETs. New wireless communication technologies are expected to significantly influence the design and implementation of MANETs in the military environment. Since the future technology combining wireless local networks and cellular networks is more and more being referred to, and defined as the fourth generation (4G) of communication systems, it is critical to understand the meaning of 4G and its Potential in influencing wireless networks, particularly MANET.

This paper is organized in the following way: Section 1 introduces the different types of wireless mobile generations. Section 2 presents Applications of the 4G design. The following sections describe the definition of 4G as a significant factor influencing wireless networks. Section 3 details how 4G technology might influence networks. Section 4 highlights security issues of 4G, section 5 describes the Quality of Service in 4G. Finally, Section 6 concludes and describes future work.

2. Applications of 4G

With the increase in the data rates, the mobile phones are made to perform higher performance applications. In 4G the mobile phone is not only for calling but its something extraordinary device that can be used for variety of purposes. One such application in 4G is context awareness. For example if the mobile user is passing by an office where he/she is having an appointment to meet someone and they have forgotten the appointment. If the office location, address and geographical location matches the one user has already stored in the phone, he/she will receive information about the appointment and will be reminded that you need to perform this activity. Telemedicine is another application of 4G [8]. Using telemedicine a patient can send general reading like temperature, glucose level and blood pressure to the doctor online. Or if someone needs to know about their family member's health continuously they can receive all the information through telemedicine by using 4G technology.

LTE

Long Term Evolution is an emerging technology for higher data rates. It is also referred as 3.9 G or super 3G technology. LTE is developed as an improvement to Universal Mobile Telecommunication System by

3G Generation Partnership Project (3GPP). LTE uses Orthogonal Frequency Division Multiple Access (OFDMA). The download rate in LTE is 150 Mbps and it utilizes the available spectrum in a very sophisticated way. In LTE the IP packet delay is less than 5 mille seconds which provides the experience of wired broadband internet access in wireless environment. The mobile TV broadcast is facilitated by LTE over LTE network.

3. Networks of 4G

Although there are different ideas leading towards 4G, some concept and network components frequently come up as a supporting and significant solutions that help achieve progress towards 4G. In this section we are going to investigate and explain technological innovations such as MIMO (Multiple-Input Multiple-Output), OFDMA (Orthogonal Frequency Division Multiple Access)
That could significantly increase security, mobility and throughput of 4G.

3.1 MIMO

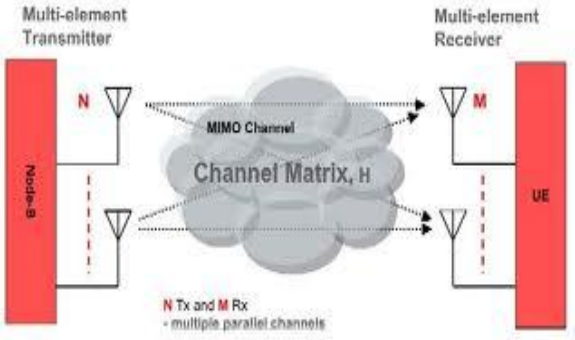

Figure 3 MIMO works

Imagine that you can hear better what you want to listen, and don't hear what bothers you, just by pointing out where message and noises are coming from. Beam forming that is the significant concept of MIMO (Multiple-Input Multiple-Output) allows you do just that using smart antennas system. But that's not all what MIMO has to offer. Spatial multiplexing, achieved by independent simultaneously working antennas, increases bandwidth Capacity by modulating and transmitting signal through many paths. Using space-time coding, reliability is improved. MIMO achieves great success thanks to multiple antennas that allows simultaneous directional transmission of two or more unique data streams sharing the same channel. Increasing speed and range, MIMO is already accepted by researchers as one of the main components of projects such as WiBro, WiMAX, WLAN, 802.11n, UMTS R8 LTE, and UMB.

3.2 OFDMA Evolution

Data comm. Research Company proposed the simplest way to implement MIMO is by sharing frequency using OFDM, that together significantly can increase performance by extending range, boosting speed and Improving reliability. Together with MIMO, OFDMA is another component of 4G that as the alternative to CDMA, promises high data capacity and spectral efficiency. OFDM (Orthogonal Frequency-Division Multiplexing) is the modulation scheme which divides allocated frequency channel into many narrow bands guaranteeing mutual independency between subcarriers such that there is no interference between them; signals are orthogonal.

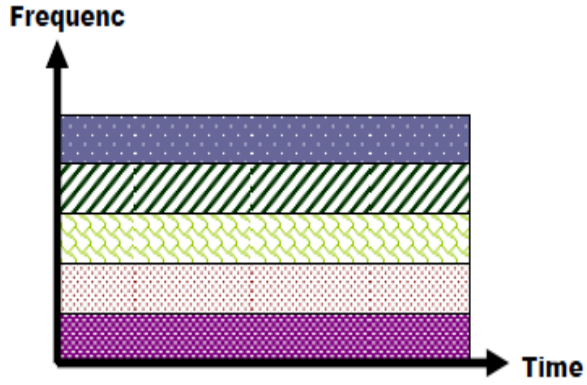

Figure 4 **FDMA**

Before OFDM become popular, three other solutions were presented to share radio spectrum between multiple users. Used by 1G, FDMA (Frequency Division Multiple Access,
Fig. 4) partitioned channel between users. To increase channel capacity, TDMA (Time Division Multiple Access, Fig. 5) was proposed that allocate each user access to the whole bandwidth for short period of time.

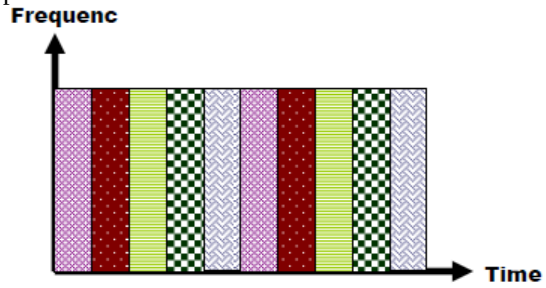

Figure 5 **TDMA**

Because TDMA in 2G caused interference problem, CDMA (Code Division Multiple Access) was

proposed as a new form of multiplexing where each user was allowed to use the whole channel capacity all the time. Different messages were transmitted with associated special code which was later used to distinguish between them.

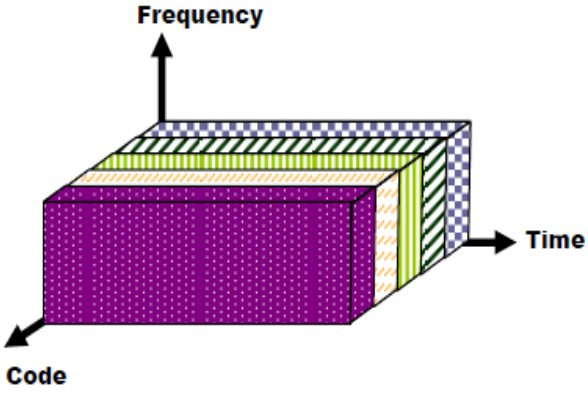

Figure 6 CDMA

The better data throughput become possible, when using orthogonal frequency division multiplexing, the radio bandwidth could be subdivided into narrow bands.

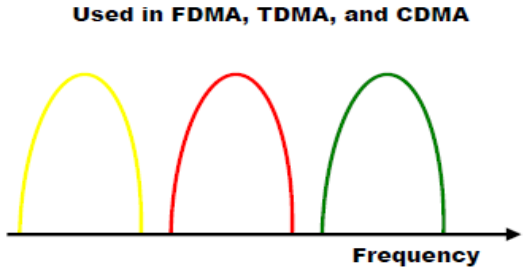

Figure 7Sharing frequency in FDMA, TDMA, and CDMA

4. Security

Security in digital world means to protect the digital systems from criminal and unauthorized usage. In terms of computers and mobile communications the need for security has increased overwhelmingly with the improvement in technology. Some decades ago when first generation of mobile networks were in use the concept of security was not so much in practice or we can say that awareness was not that much highlighted. But as technology kept on improving and new advents were introduced the need of security kept on creeping. These days no one likes to be insecure digitally. Because of the heavy dependence on digital media for the use of private, sensitive, financial and important communication. There can be many attacks on digital data some of them are eavesdropping, man in the middle attack, denial of service (DOS) attack, spoofing and lot more.

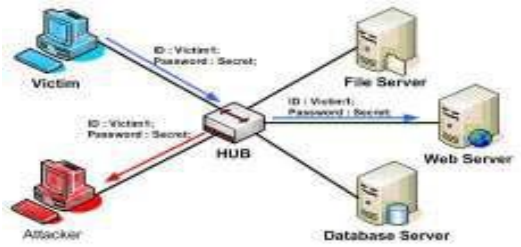

Figure 8 Eavesdropping

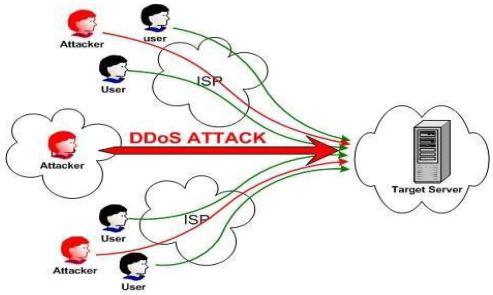

Figure 9 Denial of Service

Traditionally network security is considered to secure network edges from external attacks. Unfortunately this is not sufficient as attackers look for breaches in network protocols, traditionally network security is considered to secure network edges from external attacks. Unfortunately this is not sufficient as attackers look for breaches in network protocols, operating systems and applications. Therefore we need a comprehensive security mechanism that can protect the whole network. We can design security architecture on the basis of following objectives:

Availability: keeping the network and its components secure from malicious attacks so that there is no break during service flow.

Interoperability: using security solutions that are applicable to most of the 4G applications. They should avoid interoperability issues.

Usability: end user shall use the security mechanism easily.

QoS: security solutions should follow QoS metrics. Cryptographic algorithms used for voice and multimedia shall meet QoS constraints.

Cost effectiveness: security mechanism should cost as less as possible.

Third generation of networks provide voice and paging service to facilitate interactive multimedia. The applications include teleconferencing, internet access, video streaming, multimedia messaging and so many others. In fact 3G provide a launching pad for applications such as wireless web, email (SMS, MMS), multimedia services like video streaming etc. Fourth generation is to address the future uses of the customers in terms of higher data rates and increased bandwidth utilization. 4G is built on the concept of IP core

accommodating various heterogeneous networks. In fact 4G acts as a platform for heterogeneous networks. A service subscriber uses one of the access networks providing service from one platform. This openness and flexibility increase the probability of security breach in one of the main components of the system. Therefore the need for security has become more dominant because of the nature of the participating networks.

5. Quality of Service in 4G

In telecommunication the term QoS (Quality of Service) stands for the resource reservation control mechanism, instead of the translation of term as achieved service quality. Communication occurs when the data flows from source to destination and QoS guarantees a specified level of bit rate, jitter, and delay and packet drop probability to the flow. QoS assurance is important for real time traffics like Voice over IP (VoIP), online gaming, IP TV and video streaming etc. QoS enables network administrators to avoid network congestion and manage the network resources efficiently.

The goal of the 4G is to provide the users the facility of Always Best Connected (ABC concept). Fourth generation of networks is a combination of different networks. It gives a platform for various technologies to be accessed. To provide QoS in 4G is not simple and easy job as one has to deal with different parameters in different technologies. Like if a user is moving and changing his coverage network, so to provide service under QoS framework is challenging. While a mobile user is moving from one network to another network his communication session needs to be maintained seamlessly irrelevant of the coverage network. Similar is the case with video conferencing and video streaming, the users like to receive the services seamlessly.

There are some protocols designed to maintain the seamless communication of the users while moving or in other words to minimize the latency and packet loss of the ongoing communication session. The mobility protocols are Mobile IPv6, Hierarchical MIPv6, Fast MIPv6 and some more (details of all these protocols are given in chapter Handovers). These protocols can help in improving the mobility management of mobile users. In order to provide QoS to the mobile users we propose a combination of mobility protocol Seamless Mobile IPv6 (SMIPv6) and Session Imitation Protocol (SIP). There are two types of losses when a mobile user switches network, one is called segment packet loss and the other is called edge packet loss. Segment packet loss is because of the undeterministic nature of the handoff .While the edge packet loss is between the Mobility Anchor Point (MAP) and the MN. To minimize these losses different approaches are used, to minimize edge packet loss the MN is moved as close to the MAP as possible, while for the segmented packet loss two approaches are used one is synchronized packet simulcast (SPS) and hybrid simulcast mechanism are used. In SPS the packets are sent to both the current network as well the potential network the MN is approaching [14]. While hybrid simulcast mean that the mobile node informs the network about the handoff to be taken into effect but it is decided by the network to which AR the MN shall attach. This way the packet loss is minimized (the detailed mechanism is given in chapter of handover). Session Initiation Protocol (SIP) is used to manage mobility of different entities such as session, terminal, service and personal mobility. It facilitates mobility and maintains the real time multimedia sessions. SIP is an application layer protocol therefore it can work both in IPv4 and IPv6. SIP work along with other protocols Such as Real Time Transport Protocol (RTP).

6. Conclusion

In this paper we are describing about the various wireless mobile technologies, and various applications of 4G mobile communication as well as the LTE (Long Term Evolution). And also we describe about various networks we are used in 4G, such as MIMO and OFDMA Evolution, in that we discus about FDMA, CDMA, as well as TDMA. And also describes the Security, Quality of Service in 4G. We present the challenges that 4G faces and their up-to-date solutions. To improve the QoS in 4G we propose our own scheme of combining mobility protocol SMIP and application layer protocol SIP. With this scheme the QoS level in 4G can be improved because both the protocols provide support in handovers. Together they can decrease the packet loss and can improve security during the handover process. We can make sure the resource allocation during the handover process by combining the two protocols and mobility management can be optimized.

Future work

In future work we suggest that SIP could be combined with other mobility protocols to facilitate the mobility management and improve QoS in 4G networks.

References

1. http://www.mobile-phone- directory.org. Visited 09, February 2010.
2. http://en.wikipedia.org/wiki/2.5G. Visited 10 February 2010.
3.http://www.mobileinfo.com/3G/3G_Wireless.htm. Visited10, February 2010.
4Http://en.wikipedia.org/wiki/ 3G. Visited11 February 2010.
5.http://techcrunchies.com/3g-subscription-penetration-worldwide/. Visited 11, February 2010.
6. 4G wireless technology: when will it happen? What will it offer? Krenik, B.; Solid-State Circuits Conference, 2008. A-SSCC '08. IEEE Asian Digital Object Identifier:10.1109/ ASSCC.2008.4708715 Publication Year: 2008 , Page(s): 141 – 144
7. Research on coexistence of WiMAX and WCDMA systems, Zheng Ruiming; Zhang Xin; Li Xi; Pan Qun; Fang Yinglong; Yang Dacheng; Vehicular TechnologyConference Fall (VTC 2009-Fall), 2009 IEEE 70th Digital Object Identifier:10.1109/VETECF. 2009. 5378806 Publication Year: 2009
8.http://www.ericsson.com/thecompany/press /releases/2010/01/1372929 visited 25, February 2010.
9. A Survey of Security Threats on 4G Networks, Yongsuk Park; Taejoon Park; Globecom Workshops, 2007 IEEE Digital Object identifier:10.1109/GLOCOMW. 2007.4437813 Publication Year: 2007 , Page(s): 1 - 6
10. Integrating fast mobile IPv6 and SIP in 4G network for real-time mobility, Nursimloo, D.S.; Chan, H.A.; Networks, 2005. Jointly held with the 2005 IEEE 7th Malaysia International Conference on Communication. 2005 13th IEEE International Conference on Volume: 2 Digital Object Identifier: 10.1109/ICON.2005.1635641 Publication Year: 2005
11. "Ericsson Demos Live LTE at 144Mbps", ABI Research Wireless Daily Newsletter, February 09, 2007

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Parameter Optimization of Quantum Well Nanostructure: A PSO and GA Based Comparative Study

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***Abstract*:** Carrier transport properties of nanodevices are controlled by biasing field, frequency of the applied field and system parameters like lattice temperature, quantum well width, spacer width and carrier concentration. All these parameters are related in such a way that it is very difficult to predict optimized system parameters for desired electrical characteristics using the traditional mathematical optimization techniques. Evolutionary algorithms are stochastic methods that mimic the natural biological evolution or the social behaviour of species. Such algorithms have been used for large-scale optimization problems in many applications. In this work, two evolutionary algorithms, Genetic Algorithm (GA) and Particle Swarm Optimization (PSO), are applied to get optimized system parameters for $Al_xGa_{1-x}As/GaAs$ quantum well nanostructure, which may be utilized during fabrication for better nanodevices. The results obtained are compared in terms of convergence speed, processing time and quality of results. The PSO based algorithm is found to convergence faster than GA for almost same quality of results. And also the processing time is faster in case of PSO based algorithm for the present application of parameter optimization for nanodevice modeling.

***Keywords*:** Optimization, GA, PSO, quantum well, scattering mechanism, mobility, frequency response.

1. Introduction

Recent advances in crystal growth techniques like fine line lithography, metal organic chemical vapour deposition (MOCVD) and molecular beam epitaxy (MBE) have made possible the fabrication of low dimensional semiconductor structures such as quantum well, quantum wires and quantum dots [1-5]. A quantum well (QW) is formed when a thin layer of lower bandgap semiconductor is sandwiched between two layers of higher band-gap semiconductor [6-7]. In the quantum well structure, electrical and optical properties of the semiconductor are totally different from those in the bulk material due to quantum effect [8-9]. Due to modulation doping in QW structures, carriers are separated from ionized impurity thereby increasing the mobility of carrier due to reduced ionized impurity scattering. The carrier concentration in QW is high and the coulomb scattering is also reduced with sufficient thickness of spacer layer [10]. Theoretical studies of the electrical characteristics are clearly vital to understand the physics of these devices.

Electrical characteristics of the carrier in a QW are controlled by the system parameters like lattice temperature, well width, spacer width, carrier concentration, external dc biasing field and the frequency of applied ac field. All these parameters are related in such a way that it is very difficult to predict optimized values of parameter for desired electrical characteristics [11-12] using the traditional mathematical optimization techniques. Evolutionary algorithms that mimic the natural biological evolution or the social behaviour of species have been developed for fast and robust solution to complex optimization problems. Genetic algorithm (GA) is a computationally simple but powerful algorithm developed based on the principle of the 'survival of the fittest' and the natural process of evolution through reproduction [13]. Theoretically and empirically it is proven that GA provides robust search in complex spaces. As a result, GA is now finding more widespread applications in sociological, scientific and technological circles [14]. Despite its simplicity, GA may require long processing time to get a near-optimum solution.

PSO is an evolutionary computational intelligence-based technique, which was inspired by the social behaviour of bird flocking and fish schooling [15]. PSO algorithm shares many common points with GA. Both the algorithms start with a group of a randomly generated population, have fitness values to evaluate the population, searches for optima by updating generations and none of them guarantee success [15]. Each solution in PSO is a 'bird' and is referred as a 'particle', which is analogous to a chromosome in GA. As opposed to GAs, PSO does not create new birds from parents. PSO utilizes a population of particles that fly through the problem hyperspace with given velocities. The velocities of the individual particles are stochastically adjusted according to the historical best position for the particle itself and the neighbourhood best position at each iteration. Both the particle best and the neighbourhood best are derived according to a user defined fitness function. The movement of each particle naturally evolves to an optimal or near-optimal solution. The performance of PSO is not largely affected by the size and nonlinearity of the problem, and can converge to the optimal solution in many problems where most analytical methods fail to converge. It can, therefore, be effectively applied to different optimization problems. Moreover, PSO has some advantages over other similar optimization techniques such as GA, namely the following [16].

1) PSO is easier to implement and there are fewer parameters to adjust.
2) In PSO, every particle remembers its own previous best value as well as the neighbourhood best; therefore, it has a more effective memory capability than the GA.
3) PSO is more efficient in maintaining the diversity of the swarm [17], since all the particles use the information related to the most successful particle in order to improve themselves, whereas in GA, the worse solutions are discarded and only the good ones are saved; therefore, in GA the population evolves around a subset of the best individuals.

In the present work, both PSO and GA based optimization techniques are employed to determine the optimized system parameters for $Al_xGa_{1-x}As/GaAs$ quantum well nanostructure for nanodevice applications. The parameters to be optimized include lattice temperature, channel length, carrier concentration and spacer width to get the maximized values of mobility and cut-off frequency. Performance comparison of the two algorithms is then presented in terms of convergence speed, processing time and quality of results.

2. Analytical Model of Fitness Function

A square QW of $Al_xGa_{1-x}As/GaAs$ of infinite barrier height is considered. Reduced ionized impurity scattering and improved carrier concentration in the QW establish a strong electron-electron interaction which favors a heated drifted Fermi-Dirac distribution function for the carriers characterized by an electron temperature T_e, and a drifted crystal momentum p_d. In the presence of an electric field $\vec{F}$ applied parallel to the heterojunction interface, the carrier distribution function $f(\vec{K})$ can be expressed as;

$$f\left(\vec{k}\right) = f_0(E) + \frac{\hbar \mid \vec{p}_d \parallel \vec{k} \mid}{m^*}\left(-\frac{\partial f_0}{\partial E}\right)\cos\gamma \qquad (1)$$

where, $f_0(E)$ is the Fermi-Dirac distribution function for the carriers, ħ is Planck's constant divided by 2π, $\vec{k}$ is the two-dimensional wave vector of the carriers with energy E, m* is the electronic effective mass and γ is the angle between the applied electric field $\vec{F}$ and the two dimensional wave vector $\vec{k}$.

An ac electric field of magnitude F_1 with the angular frequency ω superimposed on a moderate dc bias field F_0 is assumed to act parallel to the heterojunction interface and thus the overall field is given by;

$$F = F_o + F_1 \sin\omega t \qquad (2)$$

As the electron temperature and the drift momentum depend on the field and the scattering processes, they will also have similar alternating components, generally differing in phase.

$$T_e = T_o + T_{1r}\sin\omega t + T_{1i}\cos\omega t \qquad (3)$$

$$p_d = p_o + p_{1r}\sin\omega t + p_{1i}\cos\omega t \qquad (4)$$

Where, T_0 and p_0 are the steady state parts, T_{1r} and p_{1r} are real and T_{1i} and p_{1i} are imaginary parts of T_e, and p_d respectively. The energy and momentum balance equations obeyed by the carrier can be given as;

$$\frac{ep_dF}{m^*} + \left\langle\frac{dE}{dt}\right\rangle_{scat} = \frac{d}{dt}\langle E\rangle \qquad (5)$$

and

$$eF + \left\langle\frac{dp}{dt}\right\rangle_{scat} = \frac{dp_d}{dt} \qquad (6)$$

Where $\langle dp/dt\rangle$ and $\langle dE/dt\rangle$, represents, respectively, the average momentum and energy loss due to scatterings and $\langle E\rangle$ depicts the average energy of a carrier with charge e. In the present model the effects of delta doping is included in the energy and momentum loss calculations to give more accurate results. We insert (3) and (4) in (5) and (6), retain terms up to the linear in alternating components and equate the steady parts and the coefficients of sinωt and cosωt on the two sides of the resulting equations following the procedure adopted in reference 6. For a given electric field F_o, we solve for p_o and T_o. The dc mobility μ_{dc} and ac mobility μ_{ac} are then expressed as:

$$\mu_{dc} = \frac{p_o}{m^*F_o} \qquad (7)$$

$$\mu_{ac} = \frac{\sqrt{p_{1r}^2 + p_{1i}^2}}{m^*F_1} \qquad (8)$$

The phase lag ϕ, the resulting alternating current lags behind the applied field is expressed as;

$$\phi = \tan^{-1}\left(-\frac{p_{1i}}{p_{1r}}\right) \qquad (9)$$

Equations 7, 8 and 9 are used as the fitness functions of ac mobility, dc mobility and phase angle, respectively. Detailed derivations of the fitness functions are available in the Ref. 7 and not deliberately included in the present analysis for brevity of this paper.

3. GA Based Optimization

In GA, a solution to a given optimization problem is represented in the form of a string, called 'chromosome', consisting of a set of elements, called 'genes', that hold a set of values for the optimization variables [18]. For appropriate binary representation four parameters (carrier concentration (N_{2D}), quatum well width (L_Z),lattice temperature (T_L) and spacer width (L_S)) are coded into a single finite length string of twenty three bits as shown in Fig 1.

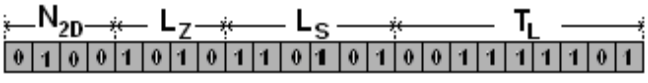

Figure 1. A chromosome with 23 bits.

To start with, a random population of chromosomes is generated. The fitness of each chromosome is determined by evaluating it against a fitness function and an average is computed which is considered as the starting average fitness value. Strings/chromosomes are then selected from the generation according to their fitness value. Strings, whose fitness value is less then the average fitness value, are rejected and will not pass to the next generation. Subsequent generations are developed by selecting pairs of parent strings from present genetic pool to produce offspring strings in the

next generation, which is called "crossover" operation. For crossover operation an integer position (t) along the string is selected randomly between 1 and the P-1, where P is the maximum string length. Two new strings are created by swapping all binary bits between positions (t + 1) and P inclusively. As an example, two consecutive strings S_k and S_{k+1} are shown in Fig. 2. A random number is chosen between 1 and 22 (23 - 1), as P = 23 here. Then the result of cross over which produces two new strings S'_k and S'_{k+1} are indicated in Fig. 2.

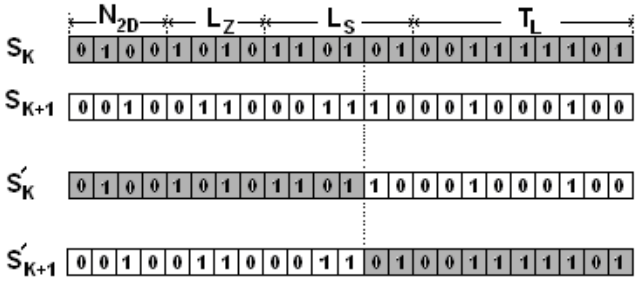

Figure 2. Cross over operation between two consecutive strings.

Gradually, generation-by-generation the algorithm will progress towards the optimum solution. When the improvement in the average fitness value falls in the range of 0.00 - 0.05% for at least ten consecutive generations, program will be terminated. The mutation operator plays a secondary role in the simple GA and mutation rates are simply small in natural population. It is also a rare process that resembles a sudden change to an offspring. This can be done by randomly selecting one chromosome from the population and then arbitrarily changing some of its information. The benefit of mutation is that it randomly introduces new genetic material to the evolutionary process thereby avoiding stagnation around local minima [19]. The four parameters that affect the performance of GA are population size, number of generations, crossover rate, and mutation rate. Larger population size and large number of generations increase the likelihood of obtaining a global optimum solution, but substantially increase processing time.

4. PSO Based Optimization

The PSO algorithm begins by initializing a group of random particles (solutions) and then searches for optima by updating generations. The fitness values of all the particles are evaluated by the fitness function to be optimized. An iterative process to improve these candidate solutions is set in motion. With the progress of 'iteration', which is synonymous to generation in case of GA, every particle updates its position, velocity and moves through the problem or solution space. In iteration, position of each particle is updated by following two "best" values. The first one is the best solution or fitness value that the particular particle has achieved so far and is called "p_{best}" and the second one is the best solution obtained by any particle in the entire population and is known as the global best or "g_{best}"[20]. After finding p_{best} and g_{best} the particle updates its velocity and positions using the following two equations:

$$\vec{v}_i(t) = w\vec{v}_i(t-1) + c_1 * rand() * (p_{best} - \vec{x}_i(t-1)) + c_2 * rand() * (g_{best} - \vec{x}_i(t-1)) \quad (10)$$

$$\vec{x}_i(t) = \vec{x}_i(t-1) + \vec{v}_i(t) \quad (11)$$

Where $\vec{v}_i(t)$ is the particle velocity, $\vec{x}_i(t)$ is the current position of the particle, w is called the inertia factor, rand () is a random number between (0,1), c_1 and c_2 are learning factors.

The following procedure can be used for implementing the PSO algorithm [21].

1) Initialize the swarm by assigning a random position in the problem hyperspace to each particle.
2) Evaluate the fitness function for each particle.
3) For each individual particle, compare the particle's fitness value with its p_{best}. If the current value is better than the p_{best} value, then set this value as the p_{best} and the current particle's position, x_i, as p_i.
4) Identify the particle that has the best fitness value. The value of its fitness function is identified as g_{best} and its position as p_g.
5) Update the velocities and positions of all the particles using equations (10) and (11).
6) Repeat steps 2–5 until a stopping criterion is met (e.g., maximum number of iterations or a sufficiently good fitness value).

5. Results and Disquisitions

The GA and PSO algorithms have been coded using MATLAB7.5 and all experiments were conducted on a 2.20GHz AMD ATHLON Processor with 2GB RAM Desktop PC. The parameters used for the algorithms are given in Table 1 and they are taken based on the consideration presented in Refs. 18 and 20.

Parameters	GA	PSO
Population Size	100	100
Max Generation/ Iteration	Varies	Varies
Selection Type	Random	NA
Crossover Rate	80%	NA
Mutation Rate	9%	NA
w_{man} ,w_{min}	NA	0.9,0.1
c_1,c_2	NA	1.49

Table 1. Parameters used for GA and PSO algorithm.

The material parameters for the $Al_{0.3}Ga_{0.7}As/GaAs$ QW are taken from Ref. 7. The range of QW parameters taken for optimization are based on theoretical assumptions and physical phenomenon and are as follows:

1. 2-D carrier concentration (N_{2D}) is $6\times10^{15}/m^2$ to $10\times10^{15}/m^2$
2. Quantum well width (L_Z) is 8nm to 12nm.
3. Spacer width (L_S) is 10nm to 50nm.
4. Lattice temperature (T_L) is 77 K to 300K.

Since the parameter optimization is to be carried out during fabrication of nanodevices (i.e real time application), an algorithm with high average performance is the best option [22]. Therefore, GA and PSO algorithms are compared based on the Mean Best Fitness measure (MBF)/average fitness value obtained over 300 runs for ac mobility and 400 runs for dc mobility and cut-off frequency.

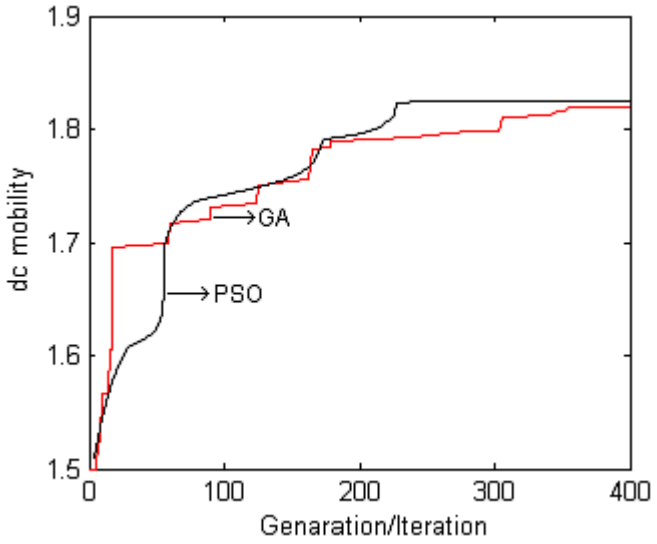

Figure 3. Plot of average dc mobility with iterations (PSO) /generations(GA).

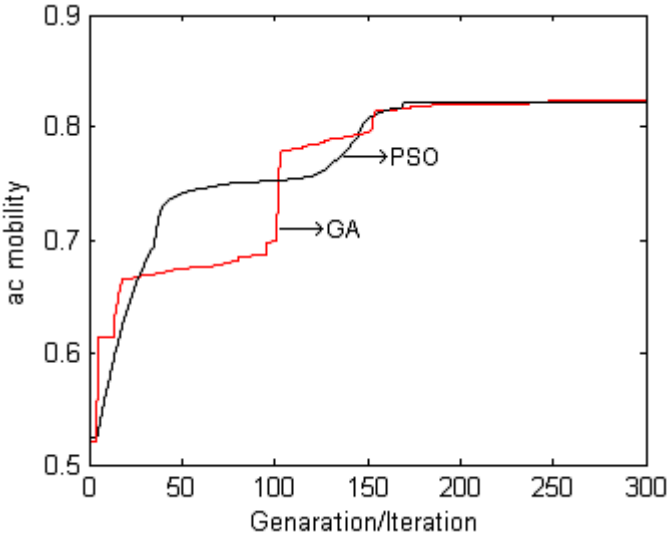

Figure 4. Plot of ac mobility with iterations (PSO) /generations(GA).

Using Eqs. 7 and 8 as the fitness functions, the GA and PSO algorithms are applied to get the optimized values of dc and ac mobility. The simulation of the search space is depicted in Figs. 3 and 4. It is found that the PSO based algorithm converges faster than GA based algorithm. For dc mobility optimization, PSO took 230 iterations and 81.23 seconds to converge whereas GA took 350 generations and 124.41 seconds. For ac mobility optimization, it was found that 165 iterations and 62.03 seconds were required by PSO and whereas it was 292 generations and 98.11 seconds for GA.
GA and PSO algorithms are applied using equation 9 as the fitness function to get the optimized value of cut-off frequency. The simulation of the search space is depicted in Fig. 5.

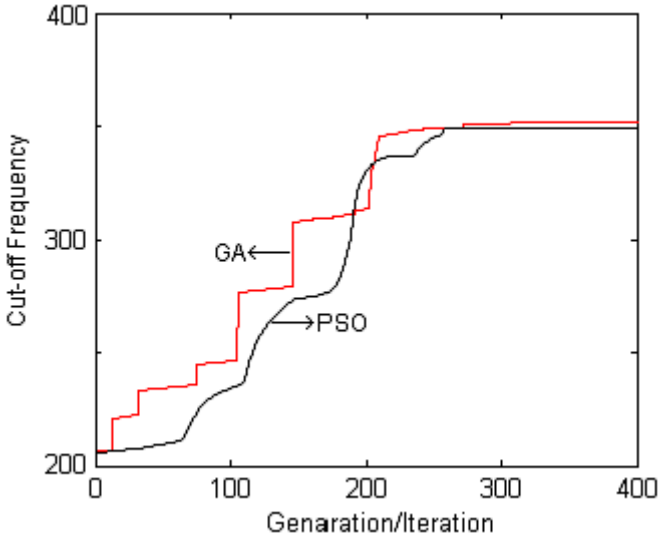

Figure 5. Plot of cut-off frequency with iterations (PSO) /generations(GA).

As in the case of mobility mobilization, it is found that the PSO based algorithm converges faster than GA based algorithm. The results obtained from using GA and PSO are summarized in Table 2.

Table 2. Performance comparison of GA and PSO.

Scheme	Convergence Time	Processing Time	Parameter value				
			MBF	N_{2D}	Lz	L_S	T_L
dc mobility optimization							
GA	350 Generations	124.4	1.821	10	12	34	80
PSO	230 Iterations	81.23	1.818	9.8	11.8	35.6	78.8
ac mobility optimization							
GA	292 Generations	98.11	0.818	10	11	31	279.5
PSO	165 Iterations	62.03	0.820	9.9	10.9	31	278
Cut-off frequency optimization							
GA	305 Generations	103.1	353	6	8	23	296
PSO	262 Iterations	93.03	350	6.3	8.5	21	293.8

AMPT: Processing time for a single run of analytical model with given parameters.

The performance of the algorithms was compared using three criteria: (1) convergence speed; (2) processing time to reach the optimum value and (3) the quality of results. The processing time, and not the number of iteration/generation cycles, was used to measure the speed of each algorithm, because the number of generations is different from one algorithm to another. To make processing time comparison more relevant and processing system independent, processing time of two proposed schemes are compared in terms of analytical model processing time (*AMPT)*.

Both the algorithms perform well at finding optimal solution. Therefore, in terms of quality of solution there seems to be no difference to distinguish GA and PSO. However, when the number of generations or iterations is taken into account, there are differences in the number taken to obtain the optimal solution. The PSO based algorithm is shown to convergence faster than the GA based algorithm. And also the processing time is less in case of PSO based algorithm for the present application of parameter optimization for nanodevice modeling.

5. Conclusion

Application of soft-computing tool, especially PSO, for parameter optimization in quantum well structure is new and quite useful to understand optimum combinations of parameters to get better quantum well nanostructure. Under similar software and hardware environment, PSO and GA is applied for parameter optimization of $Al_xGa_{1-x}As/GaAs$ QW nanostructure and performance of the two schemes are compared in terms of convergence speed, processing time and quality of results. The particle swarm optimization based algorithm is found to convergence faster than GA for almost same quality of results. Another reason why PSO based algorithm is attractive is that there are few parameters to adjust making it much simpler to implement. Through this computational study, mobility (ac and dc) and cut-off frequency values are optimized with respect to parameters of quantum well nanostructure, which will provide valuable information for the technologists involved in the fabrication of QW nanodevices. Successful implementation of such types of soft computing tools for parameter optimization of QW revels that those schemes can be successfully implemented for parameter optimization of other nanostructures.

References

[1] S. K. Sarkar, D. Chattopadhyay, One dimensional warm electron transport in GaN quantum well wires at low temperature, Phys. Rev., 6 (2000) 264-268.

[2] R. Akimoto, B.S. Li, K. Akita, T. Hasama, Sub-picosecond saturation of intersubband absorption in (CdS/ZnSe)/BeTe quantum-well waveguides at telecommunication wavelength, Appl. Phys. Lett., 87 (2005) 181104.

[3] C. Fujihashi, T. Yukiya, A. Asenov, Electron and Hole Current Characteristics of n-i-p-Type Semiconductor Quantum Dot Transistor, IEEE Transactions on Nanotechnology, 6(2007)320 – 327.

[4] G. Dewey, M. K. Hudait, K. Lee, R. Pillarisetty, W. Rachmady, M. Radosavljevic, T. Rakshit, R. Chau, Carrier Transport in High-Mobility III–V Quantum-Well Transistors and Performance Impact for High-Speed Low-Power Logic Applications, IEEE Electron Device Letters, 29 (2008) 1094.

[5] C. Weisbuch, B. Vinter, Quantum Semiconductor Structures: Fundamentals and applications, Academic Press, New York, 1991.

[6] D. Chattopadhyay, Two-dimensional electronic transport in $In_{0.53}Ga_{0.47}As$ quantum Wells, Appl. Phys. A, 53 (1991) 35.

[7] S. K. Sarkar, P. K. Ghosh, D. Chattopadhyay, Calculation of high frequency response of two dimensional hot electron in GaAs quantum well, J. Appl.Phys. 78 (1995) 283-287.

[8] A. Gold, Mobility of thin AlAs quantum wells: Theory compared to experiment, Appl. Phys. Lett., 92 (2008) 082111.

[9] Kevin F. Brennan and April S. Brown, Theory of Modern Electronic Semiconductor Devices, John Wiley & Sons, 2002.

[10] L.D. Nguyen, Ultra-High-Speed Modulation-Doped Field-Effect Transistors: A Tutorial Review, Proceedings of the IEEE, 80 (1992) 494-518.

[11] Y.F. Yang, C.C. Hsu, E.S. Yang, Integeration of GaInP/GaAs heterojunction bipolar transistor and high electron mobility, IEEE Electron Device Letters,17 (1996) 363–365.
[12] C . Weisbuch and B. Vinter, Quantum Semiconductor Structures: Fundamentals and applications, Academic Press, New York, 1991.
[13] Holland J., Adaptation in natural and artificial systems. Ann Arbor, MI: University of Michigan Press, 1975.
[14] R. S. Zebulum, M. S. Vellasco, M. A. Pacheco, Variable Length Representation in Evolutionary Electronics Evolutionary Computation, MIT Press, 8(2000) 93 -120.
[15] J. Kennedy, R. Eberhart, Swarm intelligence, Morgan Kaufmann Publishers, San Francisco, CA, 2001.
[16] Y. D. Valle, G. K. Venayagamoorthy, S. Mohagheghi, J. C. Hernandez, R. G. Harley, Particle swarm optimization: Basic concepts, variants and applications in power system, IEEE Trans. on Evolutionary Computation, Vol. 2 (2) (2008) 171-195
[17] A. P. Engelbrecht, Particle swarm optimization: Where does it belong ?, in Proc. IEEE Swarm Intell. Symp., (2006) 48–54
[18] Goldberg DE., Genetic algorithms in search, optimization and machine learning. Reading, MA: Addison-Wesley Publishing Co, 1989
[19] Subir Kumar Sarkar , Multiple lavel optimization of high frequency ac mobility in GaAs quantum wells under hot electron condition, Elsevier, 29 (2004) 243-249.
[20] Nadia Nedjah, Luiza de Macedo Mourelle, Swarm Intelligent Systems, Springer-Verlag Berlin Heidelberg 2006
[21] Y. Shi, Feature article on particle swarm optimization, IEEE NeuralNetwork Society, Feature Article, (2004) 8–13.
[22] A.E. Eiben, M. Jelasity, A critical note on experimental research methodology in EC, WCCI, vol. 1 pp. 582-587, Computational Intelligence, WCCI, Proc. of the 2002 Congress on Evolutionary Computing (CEC'2002)

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He served Oil and Natural Gas Commission (ONGC) as an Executive Engineer for about 10 years (1982 to 1992) before coming to teaching profession. He joined as a faculty member in the Dept. of Electronics and Telecommunication Engineering, Bengal Engineering and Science University, Shibpur in April 1992 (from 1992 to 1999). In 1999 he joined in Jadavpur University in the same dept. where he is presently a Professor. He has developed several short courses for the needs of the Engineers. He has published three Engineering text books and more than 260 technical research papers in archival journals and peer – reviewed conferences. Thirteen students have been awarded PhD (Engg) degree under his guidance and eight more are presently perusing PhD under his guidance. He is on the TCP of several major international/ national conferences/ symposiums and was the organizing Secretary of International conferences on communications, devices and intelligent systems (CODIS2004). As Principal Investigator he has successfully completed four R&D projects and two more are running. He has visited several countries like France, UK, Switzerland and Japan for Technical reasons like training, presenting papers or visiting sophisticated laboratories. He was invited as a post doctoral Research scholar in the Dept. of Electrical and Computer Engineering, Virginia Commonwealth University to work on quantum computing, spintronics, quantum dot memory and electronic transport in Nanostructures. His most recent research focus is in the areas of simulations of nanodevice models, transport phenomenon, single electron & spintronics devices and their applications in VLSI circuits, low power VLSI design, ad hoc wireless networks, wireless mobile communication and watermarking. He is a life Fellow of the Institution of Engineers and Institution of Electronics and Telecommunication Engineers, life member of Indian Association for the cultivation of Science.

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Experimental Protocol Development for a Passive Thermal Management System

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Abstract: Techniques to reduce the increasing energy costs have become a necessity for homeowners. For a typical residence, heating and cooling are two of the major energy consuming sources. A new technology was designed and developed in the Center for Industrial Research Applications at West Virginia University which utilizes passive convective cooling to help reduce this energy usage. The experimental testing protocol was developed for this novel passive thermal management system based upon typical materials and techniques used in the construction of residential homes. The base construction for the two units was identical. Each unit was equipped with 13 temperature sensors in identical positions on the shingled roof and inside the attic space. Each sensor was connected to a data logger installed in each unit. Baseline data was established in both controlled and environmentally exposed environments. The testing protocol and baseline data for both units are presented in this paper.

Keywords: Convective Cooling, Energy Conservation, Buoyancy

1 Introduction

The impact of rising energy prices on household budgets and the overall economy has increasingly become a focal point of public concern. A significant portion of the total energy costs can be attributed to heating and cooling. In 2005, approximately 111 million homes contained at least one heating unit with 65% of them for a single family detached dwelling [1]. For this type of home, the major heating sources are electricity and natural gas.

A typical residence has numerous energy consuming appliances and devices. Major contributors include air conditioning, refrigeration, and heating. Approximately 85% of the homes in the United States have cooling equipment, 76% of which use central air, while the remaining 24% have at least a window/wall unit [1].

With both heating and cooling being the major contributors to energy expenditures in most building structures, energy savings on these sources was the focus of this research. Currently, energy savings in a home can be accomplished in a variety of ways. One way is to replace normal window glass throughout the home with low emissivity (low-e) glass. Low-E glass has a special thin coating that will let visible light in, but helps to reduce the heat transfer between surfaces. Another alternative would include the planting of trees and/or shrubs to shade a building structure in summer and to provide a wind break in winter. The R-value of the insulation can be increased and it is also recommended that thermostats be adjusted to decrease energy usage. In addition, shading room air conditioners from direct sun will reduce their workload. Such energy savings measures will translate into reduced heating and cooling costs. [2]

The natural ventilation method that occurs in most structures takes advantage of two principles. First, as air is heated it becomes less dense and rises. Second, wind movement over and around a home creates areas of high and low pressure. If a space has high air outlets in conjunction with low inlets, ventilation occurs as the air within the space is heated, rises and escapes though the high outlets to be replaced by cooler air entering at the lower inlets. The greater the temperature differences between the outlet and inlet, the greater the ventilation rate. This is a natural result of buoyancy effects. Additionally, the structure serves as a differential pressure mechanism driven by creating a slightly higher pressure around the windward side of the structure. As the air passes over the ridge or top of the structure it creates a slightly lower than ambient pressure, the Bernoulli effect, and thus encouraging mass flow through the structure [3] - [5].

Enhancing and exploiting this buoyancy effect and taking advantage of the wind/structure interaction for convective ventilation is the goal of the current research. Traditional natural ventilation occurs in the interior of the attic. On the other hand, this research aims to enhance natural buoyancy driven convection in an exterior space immediately along the top most surface of a building to reduce attic temperatures. This could be accomplished with a variety of structures and control schemes retrofitted on top of an existing roof, or the design of a new roof structure with an air gap between the additional structure and the existing roof.

2 Experimental Setup

Two units were constructed for experimental verification testing, one for the control and the other as the experimental. Each unit was equipped with sensors to monitor the thermal attributes inside and outside of each unit, as well as the air velocity inside the stack vent. The base construction of each unit was identical with the only difference being that the second unit has the add-on roof feature. The construction of each building, as well as the experimental testing setup, is described next.

The attic test module is a gabled attic built with roof pitch of 45 degrees. The frame of the unit was constructed out of two-by-six pieces of lumber, placed 16 inches on center, and covered on the outside with plywood sheets for rigidity and strength. Tar paper was placed directly over the plywood

sheets and 1/8 inch thick asphalt shingles were layered on both sides of the roof.

Figure 1 Experimental Test Units during Construction

The exterior of the building has standard tan vinyl siding, thin (~2 mm thick), installed over the house wrap which was attached directly against the plywood. Typical R-13 fiberglass home insulation was installed in between each of the two-by-four studs with drywall installed in the interior of the unit. A layer of thin plastic was placed over the insulation to create a vapor barrier. The interior ceiling consisted of R-30 insulation between the two-by-six ceiling joists and drywall. Both the ceiling and each wall was painted white to help to seal and maintain a steady temperature inside of each building. The interior floor was constructed in a similar fashion as the exterior walls without vinyl siding on the exterior.

In addition, the roof has an overhang located on both sides to create a 12 inch soffit vent and a gable on the front side of each building. The soffit vent was constructed of one porous vent located in the middle with a solid piece on either side with the same pattern to the ends of the overhang. A 7 ft steel exterior door was installed on the front side of each building to mimic the effect of typical building openings. In addition, a hexagon-shaped gable was installed directly over the exterior door to allow access to the attic area, as shown in Figure 2. During testing, the exterior door was closed and the gable was sealed shut. In order to analyze the impact of the convective heat roof and to keep the influence of the roof between experimental and control the same, the ridge vent was excluded for this phase of this research.

Environmental monitoring was used to determine which method provided the best protection against temperature elevation caused by exposure to solar radiation. This was accomplished by using two data loggers, 44 resistance temperature detectors, and downloading the weather for the test site.

Test data collection was performed using two data loggers setup to monitor 15 and 29 temperature sensors for the 500 and 800 data units, respectively. Temperature sensors were placed in identical positions on each test unit to minimize the effects that varying the probe position could have on the test data. The test points for each unit are located twelve inches from the ridge (top), twelve inches from the attic floor (bottom), and in the middle of the top and bottom sensors (~30 inches from ridge) on both the exterior and interior of each roof surface. Each sensor was encased with a flame retardant, thermal urethane, as depicted in Figure 3 and Figure 4. Every exposed lead was also covered with this thermal urethane to negate radiation effects.

Figure 2 Exterior of the Test Units

Figure 3 Resistance Temperature Detector Secured with Thermal Urethane on the Roof

Figure 4 Resistance Temperature Detector Secured with Thermal Urethane in the Attic

The temperature sensors were monitored using two data loggers. The dataTaker® DT505 and DT800 were installed in the control and experimental units, as shown in Figure 5 and Figure 6, respectively. The 6-outlet power supply is on the far left with the data logger located in the middle. The dataTaker® DT505 setup also included an Omega® iServer™ Microserver to convert the serial DB-9 to an Ethernet RJ45 connection. The heat load from each datalogger setup was relatively small (5 W) and hence considered negligible for this research.

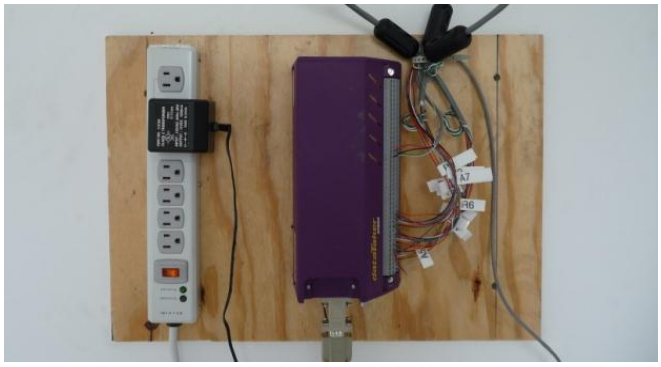

Figure 5 Data Logger Installed Inside of Experimental (800 Unit)

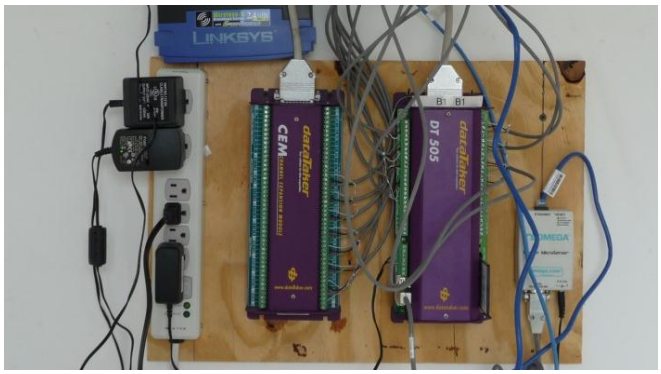

Figure 6 Data Logger Installed Inside of Control (500 Unit)

A 3-wire RTD configuration was used in the DT800 with six sensors sharing a common return to increase the amount of sensors to be measured. One lead resistance sensing loop was used for each set of six sensors necessitating each wire in the set to be the same length.

On the other hand, the dataTaker® DT505 was setup with a 4-wire configuration which used two wires to supply a constant current to the sensor while the other two wires carried no current and therefore could sense the exact voltage across the resistor without any voltage drop in the wire.

The dataTaker® software was utilized to configure, set schedules and download data from each data logger unit. The configuration included channel selection, sensor type, data logger, and label for each sensor. The labels started with the number one and were preceded with a letter for the type of location (e.g. A1 indicated attic sensor in position location one). In addition, each data logger was programmed to have Schedule A take data once a minute for each sensor. The data from each unit was downloaded at least twice per week in a comma separated values (.csv) file format over the internet or Ethernet configurations.

3 Results

Two sets of baseline data were collected for the control and experimental test units. The first set of tests was performed in a controlled environment. The units were exposed to environmental conditions for the second set of tests.

3.1 Controlled Baseline

After each unit was fully constructed, each unit remained inside of a large hangar bay, for a period of 2 months to establish baseline data in a controlled environment. During this time period, the gable was open and hence natural convective air currents would flow in through the soffit and exit through the gable. A representative set of data for the attic temperatures is shown in Figure 7 and Figure 8 for the 500 (control) and 800 (experimental) units, respectively. In addition, the same time period is shown as a representative set of data for the roof temperatures shown in Figure 9 and Figure 10 for the 500 and 800 units, respectively.

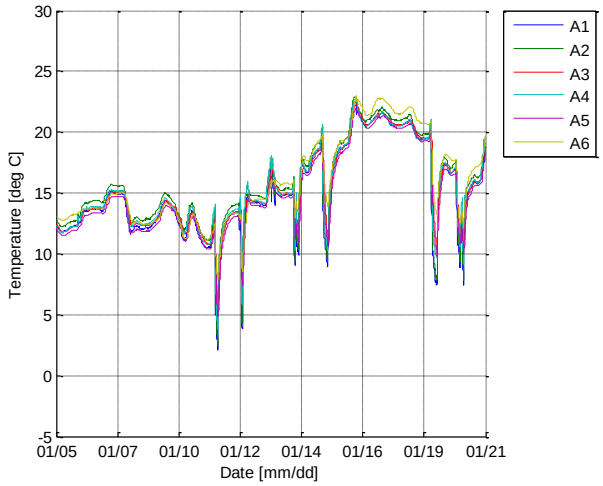

Figure 7 Controlled Baseline Attic Temperatures (500 Unit)

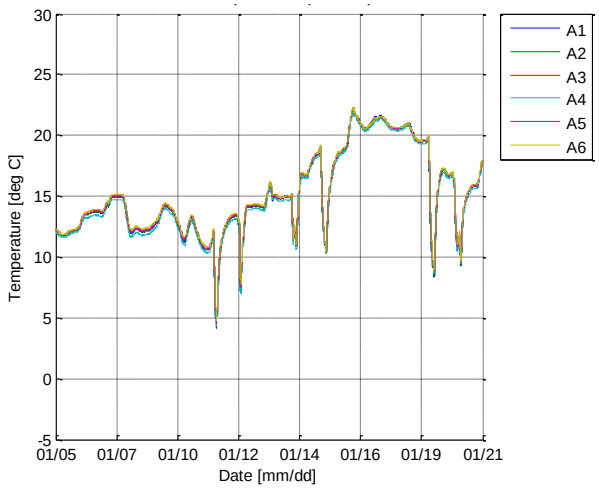

Figure 8 Controlled Baseline Attic Temperatures (800 Unit)

Each unit was located 20 feet from the hangar bay doors and raised 6 in from the floor by 6 inch square wood blocks for the controlled environment baseline testing. A series of representative tests were executed and the results were analyzed. The first test analyzed the effects of the automatic heaters located on the left and right above each of the units. The multitude of peaks in Figure 7, Figure 8, Figure 9, and Figure 10 clearly indicate when the heaters were operated intermittently.

A second test was also performed to show the opposite effect. This included fully opening the hangar bay door to allow the significantly cooler winter air to naturally flow into the hangar bay. There are six valleys shown in Figure 7, Figure 8, Figure 9, and Figure 10 on January 11th, 12th, 14th, 15th, 19th, and 20th, which clearly indicate when these events occurred. In both tests, all of the temperature sensors responded in a similar fashion in the controlled environment. The maximum difference from each of the sensors was one degree Celsius. In addition, there were minimal differences between each unit on both the roof and inside of the attic.

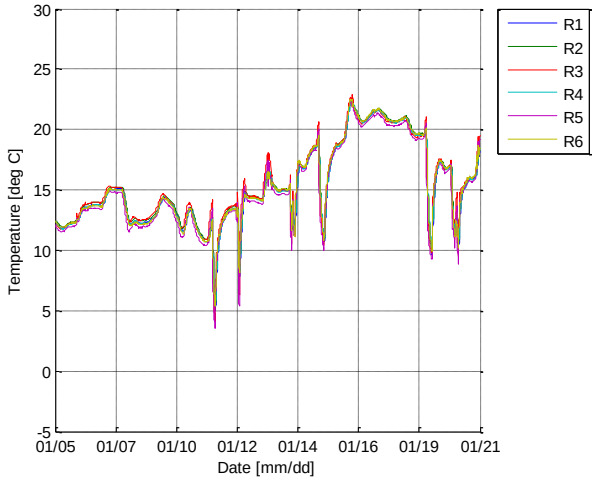

Figure 9 Controlled Baseline Roof Temperatures (500 Unit)

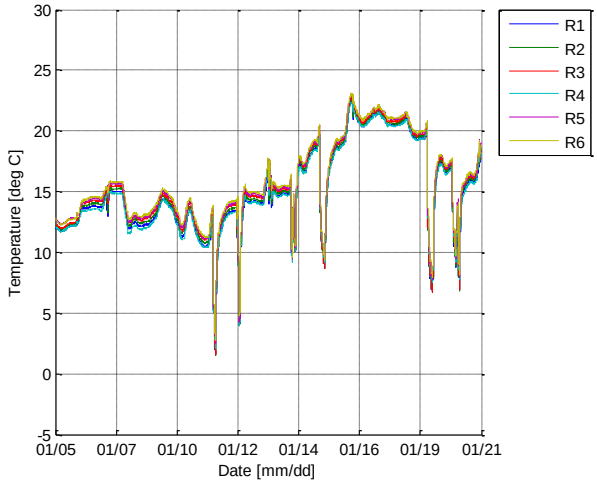

Figure 10 Controlled Baseline Roof Temperatures (800 Unit)

3.2 Environmentally Exposed Baseline

The gable on each unit was sealed with an octagon shaped particle board to minimize the convective currents in the attic. After the successful baseline testing inside of the hangar bay, each unit was moved to the tarmac area, outside of the Hangar. Testing commenced for a period of 2 months under various environmental conditions. A representative set of data for the attic temperatures are shown in Figure 11 and Figure 12 for the 500 and 800 units, respectively. In addition, the same time period is shown as a representative set of data for the roof temperatures shown in Figure 13 and Figure 14 for the 500 and 800 units, respectively.

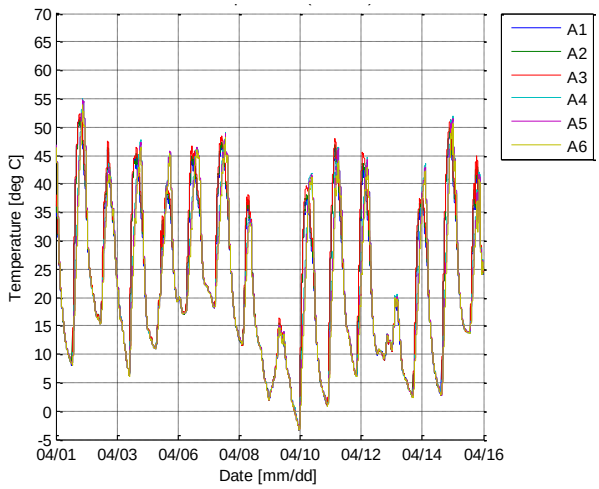

Figure 11 Environmentally Exposed Attic Temperatures (500 Unit)

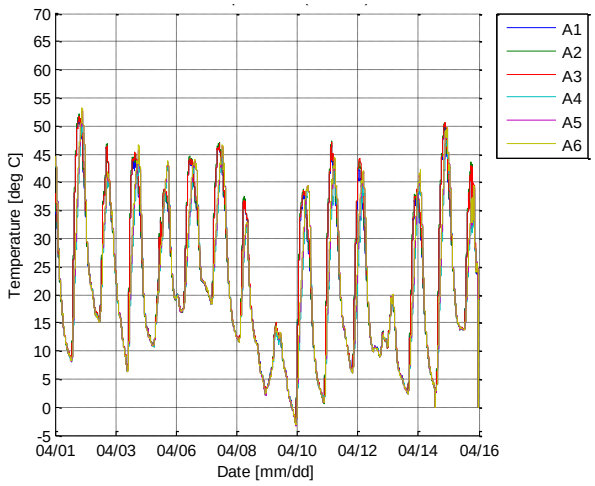

Figure 12 Environmentally Exposed Attic Temperatures (800 Unit)

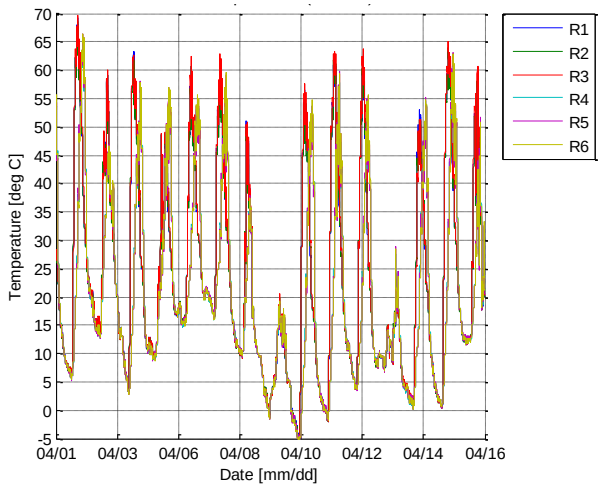

Figure 13 Environmentally Exposed Attic Temperatures (500 Unit)

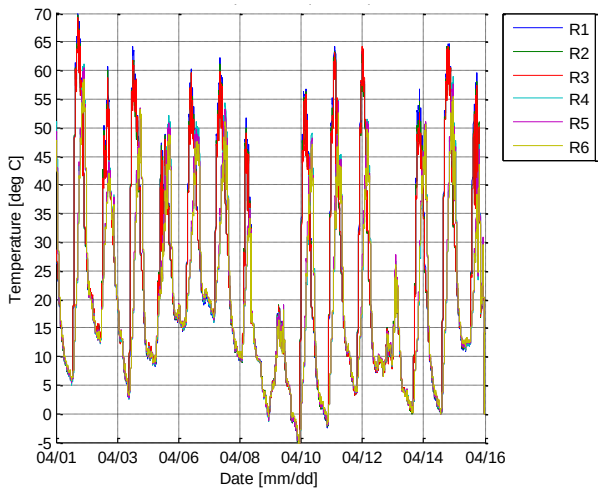

Figure 14 Environmentally Exposed Attic Temperatures (800 Unit)

Each unit was located 150 feet from the front of the hangar and placed such that they were not in the shadows of any structure for the environmentally exposed testing. The temperatures on the left hand side (LHS) and right hand side (RHS) of the roof and attic were within one degree Celsius, so an average of each side was calculated and shown in Figure 15 and Figure 16 for the 500 and 800 units, respectively. In addition, the ambient temperature for that time period is shown in each figure [6].

As expected, both sides of each unit increased as the ambient temperature increased. The LHS reached its peak temperature first, after which the RHS followed, but did not reach the same peak temperature. In this data set, there were two days (April 9th and 13th) in which the ambient temperature was significantly lower where most of the day was cloudy and hence the peak for both the RHS and LHS were similar and lower than the other testing days. In addition, the data for the LHS and RHS both follow the ambient temperature trend. For example, when there were noticeable ambient temperature fluctuations, the LHS and RHS exhibited the same pattern, as shown in Figure 17 and Figure 18, for the time period of April 13th through the April 15th when both cloudy and sunny days were observed.

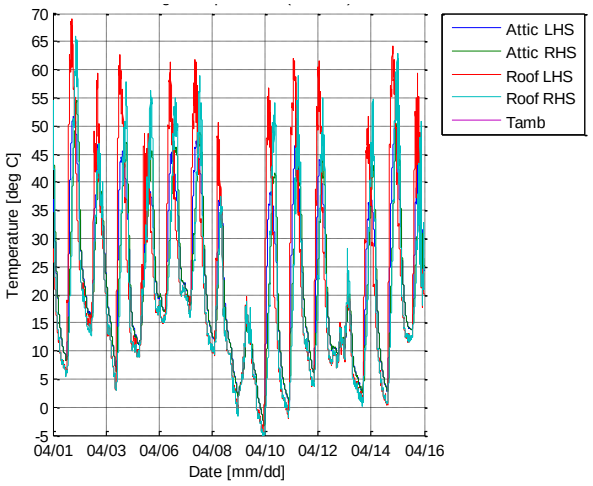

Figure 15 Baseline Average Data (500 Unit)

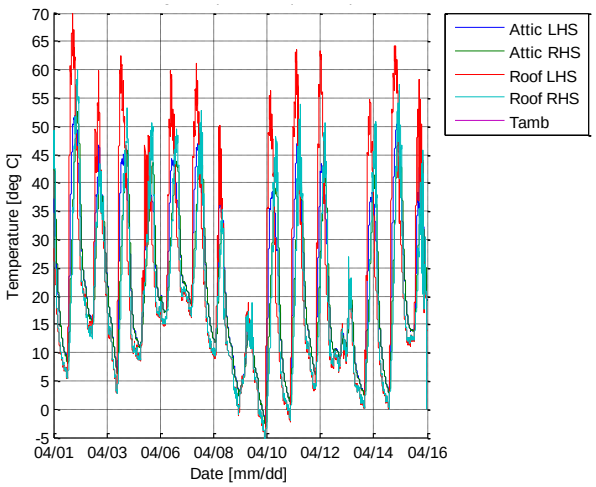

Figure 16 Baseline Average Data (800 Unit)

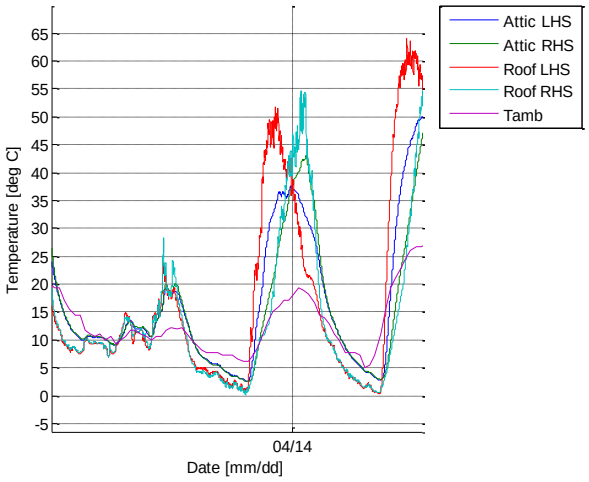

Figure 17 Environmentally Exposed Baseline Temperatures (500 Unit)

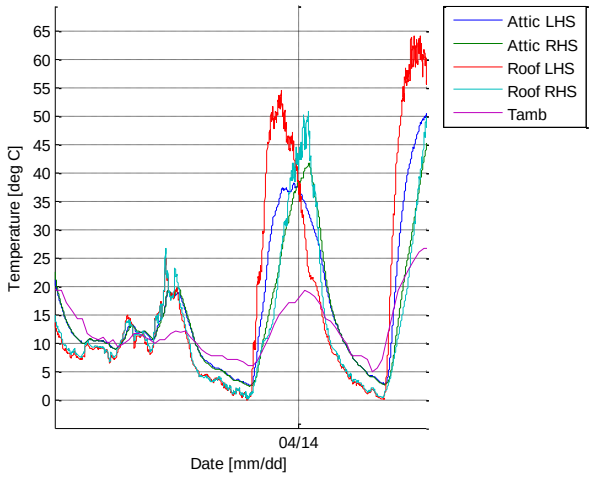

Figure 18 Environmentally Exposed Baseline Temperatures (800 Unit)

4 Conclusions

An experimental testing protocol was designed and developed for a novel thermal management system utilizing passive convective cooling. Baseline data was established in both controlled and environmentally exposed environments which confirmed each unit was built identically and can be used for further experimental testing.

References

[1] Energy Efficiency and Renewable Energy. [Online]. www1.eere.energy.gov/consumer/tips/insulation.html

[2] (2000, January) Kansas State University. [Online]. http://www.oznet.ksu.edu/

[3] Frank M. White, *Heat and Mass Transfer*. Reading, Massachusetts, 1988.

[4] Yunus A. Çengel and Robert H. Turner, *Fundamentals of Thermal-Fluid Sciences*, 2nd ed. New York, New York: McGraw-Hill, 2005.

[5] Adrian Bejan, *Convection Heat Transfer*, 3rd ed. Hoboken, New Jersey: John Wiley & Sons, Inc. , 2004.

[6] (2010) Weather. [Online]. http://www.wunderground.com

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James E. Smith received the B.S. and M.S. degrees in Aerospace Engineering and the Ph.D. degree in Mechanical Engineering from West Virginia University (WVU), Morgantown, WV, in 1972, 1974, and 1984, respectively. He is currently the Director of the Center for Industrial Research Applications at West Virginia University, where he is also a Professor in the Mechanical and Aerospace Engineering (MAE) Department. He has taught at the University since 1976, before which he was a Research Engineer for the Department of Energy (DOE). During his 30-plus-year scientific career, he has been the principal and/or co-principal investigator for various projects funded by federal agencies (TACOM, DOD, HEW, DOT, U.S. Navy, DARPA, and DOE), international corporations, and numerous U.S. corporations. The work in these projects has resulted in the publication of 164 conference papers and 50 journal or bound transaction papers. This work has resulted in the granting of 30 U.S. patents and numerous foreign patents on mechanical and energy-related devices. Dr. Smith is a member of AIAA, SAE, ASME, ISCA, ASEE, and SPIE.

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Hardware Platform for Multi-Agent System Development

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Abstract: This paper describes the conversion of WVU's analog power simulator into a micro-grid of the future test bed by installing digital relays and intelligent electronic switches. The simulator is a hardware representation of the grid which contains traditional hardware, both digital and analog, as well as the recent addition of highly connected, via Ethernet and potentially wireless communication, smart switching and monitoring devices. These new devices were chosen specifically for their cyber security capability to explore that facet of smart grid development. It is important to note that this simulator is a hardware implementation, and as such is capable of testing smart grid ideas in the most realistic setting available without affecting real customers. This simulator also has the potential to have renewable resources, like wind and solar as well as fuel cell and battery storage distributed resources tied in to test smart grid adaptability for these next generation ideas. New digital relays were installed. Microcontroller units and energy meter integrated circuits were investigated based on the desire to provide many modes of communication and as much processing power as was available in a small package. Solid state switches were designed and implemented for speed, compactness and reduced power consumption. The final configuration of this system is presented in this paper.

Keywords: Smart Grid, Power System, Hardware Prototype, Real-Time Control, Reconfigurable Power System, Multi-Agent System

1. Introduction

The United States government has recently decided to invest billions of dollars into the existing, and maybe antiquated, power grid in an effort to make it more efficient and reliable. The name given to the new power grid is the Smart Grid, the main idea being that through more intelligent devices and communication systems, both on the consumer and utilities side the grid can be made more reliable with autonomous reconfiguration and disaster mediation and more efficient with potential consumer interaction on a real time basis. This paper proposes to deliver a platform where ideas on automated reconfiguration and control via multi-agent systems (MAS's) or other control algorithms can be tested. This will be accomplished by converting a small scale analog distribution simulator using 1970's technology into a micro-grid of the future by the addition of modern digital relay equipment and micro-controlled switches.

The Lane Department of Computer Science and Electrical Engineering at West Virginia University maintains an analog power simulator that was donated to the University and installed in the 1970s. The simulator, as shown in Figure 1, the WVU power simulator in its original configuration, is a low power hardware replica of a distribution system that contains commercial, industrial and residential loads. It measures 22 feet long by 8 feet high and is 5 feet in depth. Power can be supplied to the loads from different internal and external circuits and generators and routed in a variety of ways. The simulator is, at this time, being retrofitted to represent a micro-grid of the future by installing digital relays, intelligent electronic devices, distributed energy resources (generation and storage), and potentially a FACTS device. The new digital hardware will be integrated with the older electromechanical hardware, as is found in real world power systems. This will allow the system to be used for research into autonomous reconfiguration schemes.

Figure 1: WVU power simulator in original configuration

1.1 Addition of Digital Relays

The first step in the upgrade was to install new digital relays donated by, Schweitzer Engineering Laboratories, (SEL) Inc. These are listed in table 1. The relays have been installed next to their electromechanical counter-parts and will be wired in parallel with them. This will allow either, or both, to be used if required. Figure 2 shows these new relays mounted in the simulator. Three of the relays will not be installed, the 300G generator relay and two of the 751 feeder protection relays. These will be reserved for use with the distributed generation equipment to be acquired later.

Table 1: List of equipment donated by SEL

Qty.	Relay Type
4	SEL 751A Feeder Protection Relays
1	SEL 387A Current Differential Relay
1	SEL 300G Generator Relay
4	SEL 351 Over-Current Protection Relays
1	SEL 351S Protection Relay
1	SEL 734 Revenue Metering System
1	SEL 3351 System Computing Platform with Subnet Software

Figure 2: Power simulator with added digital relays

1.2 Addition of Intelligent Electronic Switches

The simulator has approximately 100 manually operated switches that allow power to be routed over different paths between the generators and loads. The majority of those switches will be redundant with microcontroller switched optically isolated triacs. Each microcontroller will also be paired with an ADE 7758 three-phase energy metering chip to monitor current and voltage at that node and to allow the microcontroller units (MCU's) to focus on communications and higher level intelligence functions. All of the ADE7758 metering chips will be fed the same 10 MHz clock signal making it possible to synchronize waveform sampling across all nodes. When they are not needed the MCU controlled switches can be turned off and the simulator used in its original configuration.

2. Research and Development Potential

The system, with its new functionality will provide the capability to test and refine future grid handling problems and still remain familiar to the operators and designers likely to use it. The authors and their colleagues have been doing research on using distributed intelligent agents with communication capability for several years as a solution to the communications bottleneck and information overload that are fundamental problems of the present day grid. This multi-agent system (MAS) approach has been simulated in software but real world hardware with its associated timing and throughput issues is needed to really explore the concept; the upgrade to the simulator was designed with this specifically in mind. The communication requirements and processing capabilities needed for studies on MAS were at the forefront of the hardware selection process. A side benefit is that the hardware is then also applicable to other ideas using immune system based algorithms for example as intelligence models for adaptive control of the grid.

One of the first problems to be addressed by the upgraded simulator was the reconfiguration of the system in response to an outage. With the intelligent connectivity available in the upgrade there will be ways to route power around faults and restore power to much of the system. MAS algorithms [1] would have direct applicability here as would the immune based approach [2]. Once this problem has been explored and the lessons learned, it can be applied to more subtle and complicated predictive control scenarios.

With the ability to monitor load flow at virtually every node comes the ability to determine optimum power flow through the system as well as to automatically prioritize and shed load when needed or bring on distributed generation to abate a critical condition all using the distributed intelligence built into the system. These are all ideas that could be explored with the simulator that could have a tremendous impact on the grid of the future.

Communications is a critical component of any distributed intelligent system and models have shown that a multilayered communication scheme can offer a dramatic improvement in communication integrity and throughput. The MCU's chosen to act as the distributed intelligence were picked in large part for their ability to communicate via many distinct ports. Ethernet, Can bus, and RS-232 that could be used as wireless ports are all available to the algorithm explorer.

Recently Moheuddin et al. [3], published a paper on the optimization of the number and placement of distributed agents in a system that is scalable, meaning the algorithm employed to pick the placement and number was unaffected by the size of the system. Their work could be implemented and verified on the new system and could also have great impact on the evolving smart grid.

Cyber security is a very large concern in the new grid and the upgraded simulator has been designed with this in mind. A very attractive feature of these particular MCU's is that they handle cryptographic processing not with the main processor but with a dedicated cryptographic co-processor. The MCU's were designed specifically to be used in a secure mode and there is no processing capability lost by implementing state of the art security measures.

3. Intelligent Electronic Switch Hardware Selection

The digital relays bring the micro grid up to a current state of the art level, but to allow for advanced reconfiguration algorithm testing, automation of the manual switches on the simulator is desired. It would be possible to use commercial relays for this function except the size and cost of the system would be prohibitive. The solution was to design a compact solid state switch that could be controlled by a microcontroller that would also sample current and voltage at that node.

A search was made to find a compact microcontroller that gave as much processing and communication capability as currently available to provide the algorithm developers as large a canvas to try their ideas on as possible. A demo board from Freescale, the M52259DEMOKIT was chosen. It has a 32 bit 80MHZ processor with 512 Kbytes of flash memory, 64 Kbytes SRAM, and a cryptographic accelerator unit to allow for secure communications without taxing the main processor. It comes with two USB ports, one 10/100 Ethernet port, an RS-232 port and a high speed CAN bus port all in a 3 in x 3.5 in two board package. Figure 3 shows the demo board with its communication ports.

From the beginning of this project it was determined that communication was a high priority. This came directly from the desire to test MAS ideas and from the desire to have multiple communication paths to insure data integrity. The single biggest reason this particular demo board was selected was the multiple communication ports available. Both CAN bus and Ethernet can be used for a multiple wired communications scheme and the RS-232 port can then be dedicated to a wireless module. This still leaves the on board diagnostics USB port available for programming and the QSPI for interface with the ADE7758 meter chip. It is believed that the remaining USB port could be used for a memory device and that still leaves an I^2C interface available for future use.

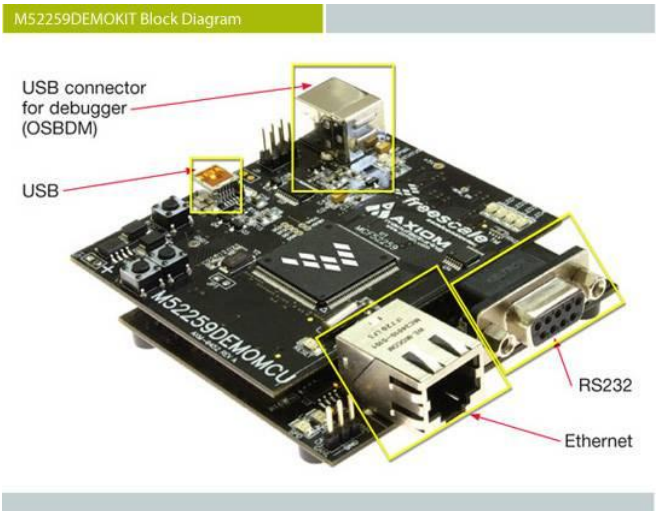

Figure 3: Freescale M52259DEMOKIT

In addition to communications it was also apparent early on that security will be a very big issue. To address this issue the processor comes with a dedicated cryptographic co processor specifically to handle secure communication.

The power simulator can have voltages as high as 750 VAC and currents in the 5 ampere range. The triacs selected are of the snubberless type and rated for 8 amps at 1000 VAC in order to give a comfortable safety margin. The triacs are driven by opto-isolators specifically designed to work with triacs. Non-zero crossing devices were chosen. Since all three phases will be switched at the same time it seemed that zero crossing switching might pose a problem during powering up.

Initially the idea was to use the A/D capability of the MCU to sample current and voltage at the node. While searching for triacs and opto-isolators a multifunction energy meter IC was discovered that would perform those functions and many more leaving the MCU with more processing and communications capabilities. The specific one chosen, the Analog Devices ADE7758, is a three phase high accuracy energy measurement IC. It has a SPI interface and a wide variety of relay type features making it an ideal if not serendipitous find. All of these parts fit together in a straightforward module as shown in the block diagram of Figure 4.

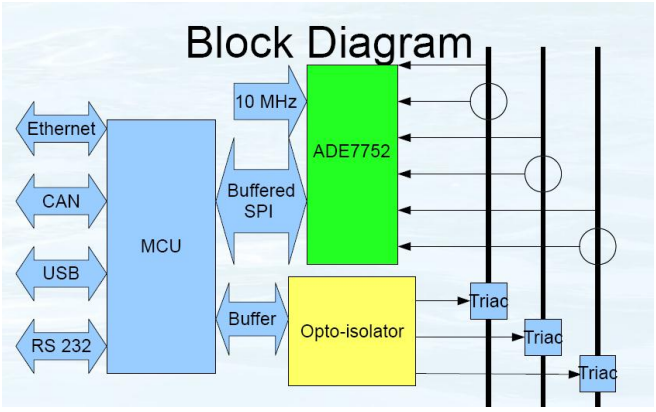

Figure 4: Block diagram of intelligent electronic switch

4. Circuit Design and Board Fabrication

The ADE7758 is only available in a surface mount package. This necessitates fabrication of a printed circuit board and surface mount soldering techniques. A design was drawn up and sent out for fabrication and a surface mount soldering station assembled and tested.

The ADE7758 Poly Phase Multifunction Energy Metering IC has built in analog to digital converters for sampling all three phases in both voltage and current. The input of the A/D's is -0.5 V to +0.5 V so the line voltages must be scaled down to that range with voltage dividers.

Traditionally, current is sensed with a current shunt of some sort often with an isolation transformer to step down the current. Effectively what this does is convert the current to a proportional voltage where that voltage is sampled and the corresponding current derived from the amplitude of the voltage. Alternatively a Hall Effect sensor could be used but they are too large for this project. The ADE7758 can be used in the normal current shunt mode but the engineers at Analog Devices have also come up with a unique alternative solution.

The voltage induced in the secondary of a transformer is proportional to dI/dT of the primary. The constant of proportionality is the mutual inductance between the primary and the secondary. Since this circuit is really only set up for signals with no DC component, the engineers decided that the secondary voltage could be sampled and then mathematically integrated to determine the current though the primary. An advantage of this is that there is almost no power that has to be dissipated in the secondary circuit.

As is common when using analog to digital conversion, anti-aliasing filters need to be used. The circuit recommended by Analog Devices was used and a copy taken directly from the data sheet as shown in Figure 5 [4].

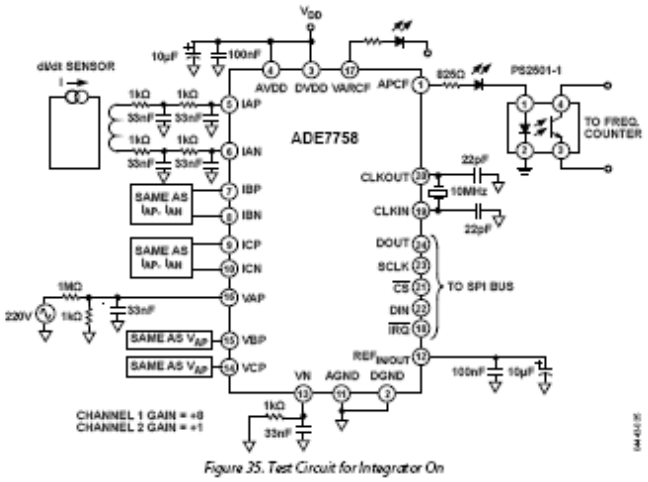

Figure 5: Circuit recommended by Analog Devices [4]

The output circuits at pins 1 and 17 were not installed nor were the capacitors for the clock circuit but the rest of the circuit was used as shown. The 5 VDC required by the IC comes directly from a regulator on the printed circuit board. This ensures that the IC receives a stable voltage. Since the M52259DEMOKIT also takes 5 VDC, they can both be supplied by the same source set so the voltage at the boards is above the 5.5V needed for ample headroom at the regulator.

The regulator chosen only supplies 50 mA but the ADE7758 only draws 21 mA maximum.

The M52259 microprocessor runs on 3.3 VDC, unfortunately the ADE7758 runs on 5 VDC causing a problem when the two are setup to talk to each other. The specifications for the input to the ADE7758 are within the output range of the M52259, so no level shifting is needed. The input to the M52259 is not specified to handle the logic levels supplied by the 5 volt ADE7758, so a level shifter 74LVC08AD was used between the two.

Only three of the four gates are needed for level shifting leaving a spare gate which can be used as a buffer between the MCU and the opto-isolators (see Figure 6). The output of the level shifter is 50 mA per gate. The opto-isolators are wired in parallel to stay under the 3 VDC supplied by the M52259DEMOKIT and draws 10 to 20 mA each. The operating load was chosen to be 20 mA so the combined draw is about 60 mA and beyond the range of the level shifter. A simple transistor circuit composed of a 2N2222 NPN small signal transistor running in saturation when on and a couple of 10 ohm current limiting resistors R1 and R2 were used as a current amplifier to ensure all parts were within their design limitations.

The triacs need a minimum gate current to turn on and that is determined by gate resistors R3 for phase A, R4 for phase B and R5 for phase C. In order to get the largest voltage range for the triac, a gate resistor was chosen that would give a bit less than the recommended maximum gate current of about 100 mA for the maximum voltage on the line. The maximum selected was 750 VAC and 75 mA as the largest RMS current desired. A 10K ohm resistor in the gate circuit then limits the gate current to about 75 mA. This should allow the triacs to work down to about 180 VAC based on the typical values for gate trigger current, but their value may have to be adjusted if the operating voltage is used in that range, because the triacs are specified to work at a 50 mA max gate trigger current which corresponds to 500 VAC.

The circuit boards were drawn up using Sunstone Circuit's PCB123® CAD software and the boards were then fabricated (see Figure 7). This was effectively a prototype so the board was designed with the ability to change the layout to accommodate minor changes. There are places in the triac section where snubber circuits can be added if they are found to be needed later. The level shifter is left disconnected with solder pads so that it can be reconfigured. Parts of the meter chip like the clock and SPI interface are left open with solder pads. With this setup, the clock signal can be supplied by an external clock or an individually dedicated one may be used.

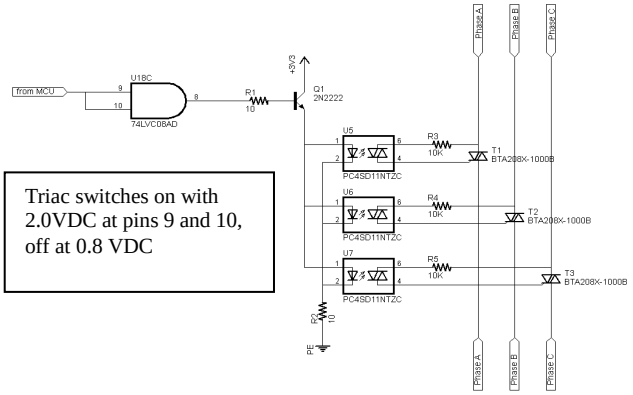

Figure 6: Triac triggering circuit

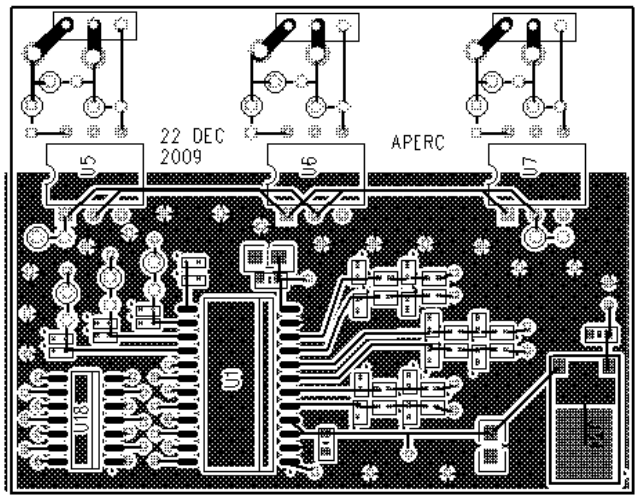

Figure 7: Sense and control PCB layout

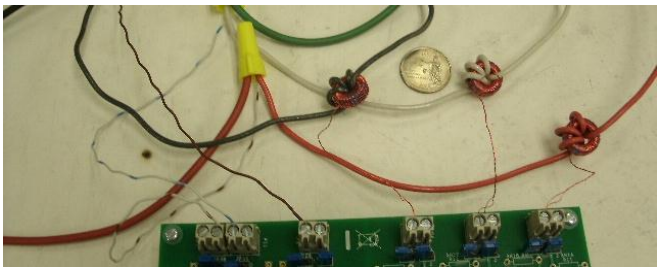

Figure 8: Current sensors being bench tested with the ADE7758 evaluation board

In an effort to minimize cost and size, the current sensors were hand wound on salvaged toroid cores for initial testing. The integration method of current measurement was chosen to keep part count down and power dissipation at a minimum since that power would have to be dissipated in the electronic switch. The sensors are simple transformers wound on the toroidial core with about 85 turns of 30 AWG as the secondary which gets attached to the ADE7758 A/D and about 3 turns of the 14 AWG that connects to the triac. This arrangement makes a very compact transformer that fits easily in the switch box and is very tolerant to noise pickup because of the toroidial configuration. The current sensors being tested are shown in Figure 8.

5. Hardware Test and Results

Since the A/D's have very high input impedance, simple voltage dividers are used for voltage sensing. The current sensor configuration chosen was novel so a bit of experimentation was required in their design. An evaluation board for the ADE7758 was purchased from Analog Devices. The board comes with LabVIEW based software with a GUI that makes it very easy to access all of the chip's capabilities. The only downside to the evaluation board is that it requires a parallel printer port for an interface to the board which is rather rare anymore. It is hoped that the source code can be modified to use an Ethernet port instead in which case the ADE7758's could be accessed though the MCU's in the intelligent switches.

To test the triacs a sample code that turns an LED on for one second then off for the next was used to do the same with

the triacs on the simulator. The entire load that was available on the simulator was switched-in gradually and the triacs performed as expected. The circuit was modified for 120 VAC by changing the gate resistors to 1.8K ohm instead of the 10K ohm that were in place for the 750 VAC of the simulator. The setup was then moved to one of the test benches where three-phase at 120 VAC was switched in the same way and the turn-on and turn-off of current and voltage waveforms were captured with a Yokogawa PZ4000 power analyzer.

The turn-on waveforms were just as expected with the addition of some noise that was not considered but can be easily explained see Figure 9. The turn-on is immediate as expected but the crossovers and distortion in the other phases at the same time are not as expected from a normal three phase power sinusoid waveform at first glance. The delay in turn-on after each cycle goes through zero is due to the finite current required through the gate to trigger the triac. Until the voltage has reached a point where that current is attained there is no conduction. The reason that the other phases are distorted during that time is that this is a three phase delta wired circuit so the sum of the currents and voltage must be zero at all times. If there is no current flow in one phase, effectively an open, the other phases will have to sum to zero regardless.

The turn-off wave forms although also not exactly what was expected makes sense in light of the previous argument, see Figure 10. The triacs lose their trigger at the mid way point of the plot and the first triac turns off, channels 5 and 6 as expected.

The turn-off with an inductive load is widely described in the literature, see Figure 11. In this case the trigger happens at the first division, or 10%. The current in each phase shuts off as expected, but the voltage in each phase tapers off very slowly. This is due to the voltage induced across the inductor and resistor in parallel as the magnetic field collapses in the inductor.

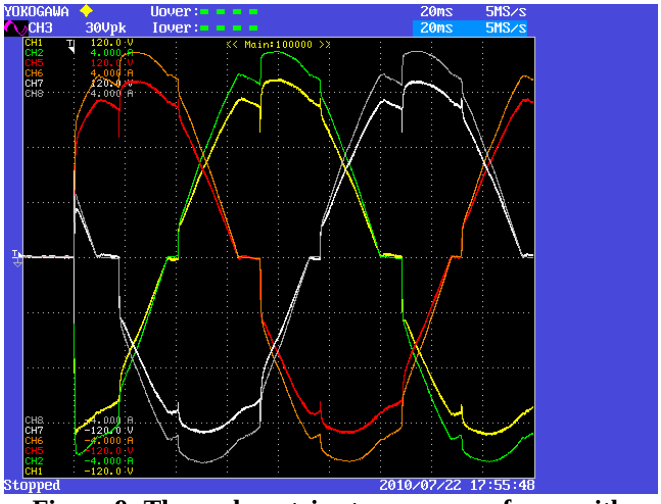

Figure 9: Three phase triac turn-on waveforms with resistive load

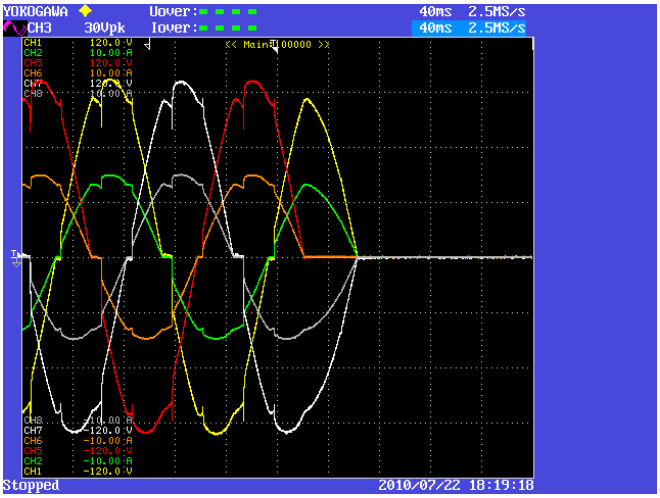

Figure 10: Three phase triac turn-off waveforms with resistive load

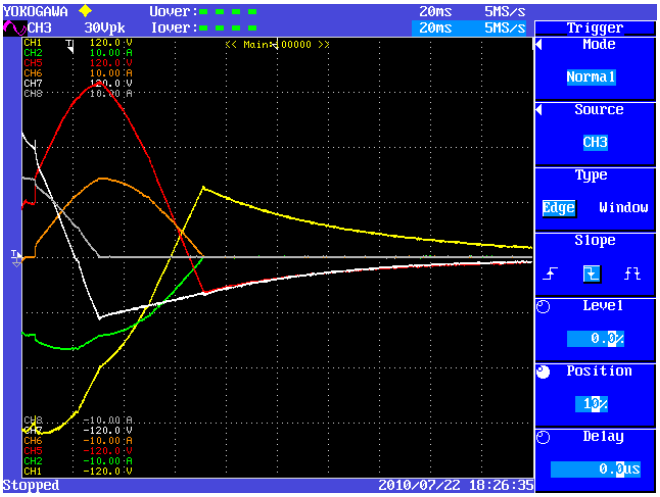

Figure 11: Three phase triac turn-off with inductive load

6. Conclusions

The sense and control boards have been designed and fabricated. A procedure is now in place for the assembly of the boards. All testing done on them has been successful. The open issues are the SPI interface and the verification of the sensor filter circuits which require some sort of communication with the ADE7758 on the board so may be best left until that issue is resolved. The 50 boards that are in house are all suitable for use. If a next version of the board is made and incorporates the new layout of the level shifter and transistor buffer it would probably save some time and make for a neater package.

The current sensors seem to work just fine, they give a voltage that's in a good range for the ADE7758 A/D's and have been shown to be acceptably linear over the range of currents expected. There may be room here for improvement with more attention paid to core selection and size. More experience with the present devices may indicate that more or less coupling would be advantageous on the power simulator. It may also be found that more than one type would give the greatest benefit. The evaluation board will be a great tool for exploring this topic in more detail

The triacs performed very well and despite the added noise induced at crossover they look to be a very satisfactory

solution as AC switches on the simulator. There is also room for experimentation and possible improvement through the design and testing of snubbers for switching more inductive loads if needed in the future, and the impact of zero crossing opto-isolators would also be an interesting topic to explore further.

References

[1] Koushaly Nareshkumar, *Application of multi-agents to power distribution systems*. Morgantown, WV: West Virginia University Libraries, 2008.

[2] Ali Feliachi Rabie Belkacemi, "Multi-Agent Design for Power Distribution System Reconfiguration Based on the Artificial Immune System Algorithm," in *ISCAS 2010*, 2010, May 30-June 2.

[3] Afzel Noore, and Muhammad Choudhry Summiya Moheuddin, "A Reconfigurable Distributed Multiagent System Optimized for Scalability," *International Journal of Computational Intelligence*, vol. 5, no. 1, pp. 60-71, 2009.

[4] Analog Devices, "ADE7758 Poly Phase Multifunction Energy Metering IC Data Sheet," 2008.

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Document Classification using Novel Self Organizing Text Classifier

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***Abstract*:** Text categorization is one of the well studied problems in data mining and information retrieval. Given a large quantity of documents in a data set where each document is associated with its corresponding category. This research proposes a novel approach for documents classification with using novel method that combined competitive self organizing neural text categorizer with new vectors that we called, string vectors. Even if the research on document categorization has been progressed very much, documents should be still encoded into numerical vectors. Such encoding so causes the two main problems: huge dimensionality and sparse distribution. Although many various feature selection methods are developed to address the first problem, but the reduced dimension remains still large. If the dimension is reduced excessively by a feature selection method, robustness of document categorization is degraded. The idea of this research as the solution to the problems is to encode the documents into string vectors and apply it to the novel competitive self organizing neural text categorizer as a string vector. The quantitative and qualitative experiment results demonstrate that this method can significantly improve the performance of documents classification.

Keywords: Text Classification (TC), Documents Classification, Information Management, Data Mining.

1. Introduction

As the volume of information continues to increase, there is growing interest in helping people better find, filter, and manage these resources. Text categorization (TC a.k.a. Text classification, or topic spotting) - the assignment of natural language documents to one or more predefined categories based on their semantic content - is an important component in many information organization and management tasks [1].

Automatic text categorization task can play an important role in a wide variety of more flexible, dynamic and personalized tasks as well: real-time sorting of email or files, document management systems, search engines, digital libraries.

In the last 10 years content-based document management tasks (collectively known as information retrieval—IR) have gained a prominent status in the information systems field, due to the increased availability of documents in digital form and the ensuing need to access them in flexible ways [2].

TC the activity of labeling natural language texts with thematic categories from a predefined set, is one such task. TC dates back to the early 60's, but only in the early 90's did it become a major sub field of the information systems discipline, thanks to increased applicative interest and to the availability of more powerful hardware. TC is now being applied in many contexts, ranging from document indexing based on a controlled vocabulary [3], to document filtering, automated meta data generation, word sense disambiguation, population of hierarchical catalogues of Web resources, and in general any application requiring document organization or selective and adaptive document dispatching.

In many contexts trained professionals are employed to categorize new items. This process is very time-consuming and costly, thus limiting its applicability. Consequently there is an increasing interest in developing technologies for automatic text categorization [4].

A number of statistical classification and machine learning techniques has been applied to text categorization, including regression models, nearest neighbor classifiers, decision trees, Bayesian classifiers, Support Vector Machines (SVM), rule learning algorithms, relevance feedback, voted classification, and neural networks.

The research on text categorization has been made very much progress in context of machine learning and data mining. It requires encoding documents into numerical vectors for using one of traditional algorithms for text categorization [5].

A corpus which is a collection of documents is mapped into a list of words as the feature candidates. Among the candidates, only some are selected as the features. For each document, a numerical value is assigned to each of the selected features, depending on the importance and presence of each feature. However, encoding documents so causes the two main problems: huge dimensionality and sparse distribution [6].

In order to solve the two main problems, this research uses the novel method that documents should be encoded into string vectors. A string vector refers to a finite set of strings which are words in context of a natural language. In numerical vectors representing documents, words are given as features, while in string vectors, words are given as feature values. Features of string vectors are defined very variously as properties of words with respect to their posting, lexical category, and statistical properties, but in this research, the highest frequent word, the second highest frequent one, and so on are defined as features of string vectors for easy implementation.

By encoding documents into string vectors, we can avoid completely the two main problems: huge dimensionality and sparse distribution.

We proposed the competitive neural text categorizer, as the approach to text categorization and proposed the application of it to documents categorization. Before creating the

proposed neural network, traditional neural networks, such as MLP (Multi Layers Perceptron) with BP (Back Propagation) receives numerical vectors as its input data. Differently from the traditional neural networks, the proposed neural network receives string vectors. It has the two layers as its architecture: the input layer and the competitive layer. It is expected for the proposed model to improve the performance of text categorization by solving the two main problems.

The rest of this paper is organized as follows. The principle of TC and previous works is given in next sections. Strategies of encoding documents were given in sections 2. Section 3 describes the novel competitive self organizing neural text categorizer model. In section 4 we will mention the simulation result and significance of this research. Conclusions are presented in Section 5.

2. Related Work

This section is concerned with previous works relevant to this research and we will survey previous relevant works, and point out their limitations. There exist other kinds of approaches to text categorization than machine learning based ones: heuristic and rule based approaches. Heuristic approaches were already applied to early commercial text categorization systems [7]. However, we count out the kind of approaches in our exploration, since they are rule of thumbs. Since rule based approaches have poor recall and require a time consuming job of building rules manually as mentioned in the previous section, they are not covered in this article, either. Therefore, this article counts only machine learning based approaches to text categorization considered as state of the art ones. Even if many machine learning approaches to text categorization already proposed, we will mention the four representative and popular approaches: KNN (K Nearest Neighbor), NB (Naive Bayes), SVM, and BP Neural Networks (NNBP or briefly BP) [8].

It requires encoding documents into numerical vectors for using one of them for text categorization; the two main problems are caused. String kernel was proposed in using the SVM for text categorization as the solution to the two main problems, but it failed to improve the performance [9]. In this section, we will explore the previous works on traditional approaches to text categorization and previous solution to the two main problems.

The KNN may be considered as a typical and popular approach to text categorization [10]. The KNN was initially created by Cover and Hart in 1967 as a genetic classification algorithm [11]. It was initially applied to text categorization by Massand et al at 1993 in [12]. The KNN algorithm is quite simple: given a test documents, and uses the categories of the K neighbors to weight the category candidates. The similarity score of each neighbor documents to test documents is used as the weight of the K nearest neighbor documents. If several of nearest neighbor share a category, then the per-neighbor weights of that category are added together, and the resulting weighted sum is used as the like hood score of that category with respect to the test document. By scoring the scores of candidate categories, a ranked list is obtained for the rest document. KNN was recommended by Yang at 1999 in [13] and by Sebastiani at 2002 in [14] as a practical approach to text categorization. Therefore, the KNN has been aimed as the base approach in other literature as the base approach. The Naive Bayes may be considered as another approach to text categorization. It was initially created by Kononenko in 1989, based on Bayes Rule [15]. Its application to text categorization was mentioned in the textbook by Mitchell in [16]. Assuming that the Naive Bayes is the popular approach, in 1999, Mladenic and Grobelink proposed and evaluated feature selection methods [17]. The Naive Bayes has been compared with other subsequent approaches in text categorization at [18].

Recently, the SVM was recommended as the practical approach to text categorization [19]. It was initially introduced in Hearst magazine in [20]. In the same year, it was applied to text categorization by Joachims [21]. Its idea is derived from a linear classifier perceptron, which is an early neural network. Since the neural network classifies objects by defining a hyper-plane as a boundary of classes, it is applicable to only linearly separable distribution of training examples. The main idea of SVM is that if a distribution of training examples is not linearly separable, these examples are mapped into another space where their distribution is linearly separable, as illustrated in the left side of figure 1. SVM optimizes the weights of the inner products of training examples and its input vector, called Lagrange multipliers [22], instead of those of its input vector, itself, as its learning process. In fact, the method is defined over a vector space where the problem is to find a decision surface that "best" separate the data points in two classes. In order to define the "best" separation, we need to introduce the "margin" between two classes. Figure 2 and 3 illustrate the idea. For simplicity, we only show a case in a two dimensional space with linearity separable data points. It was adopted as the approach to spam mail filtering as a practical instance of text categorization by Druker et al in [23]. Furthermore, the SVM is popularly used not only for text categorization tasks but also for any other pattern classification tasks [24].

In 1995, BP was initially applied to text categorization by Wiener in his master thesis [25]. He used Reuter 21578 [26] as the test bed for evaluating the approach to text categorization and shown that back propagation is better than KNN in the context of classification performance. In 2002, Gabriel applied continually BP to text categorization [27]. They used a hierarchical combination of BPs, called HME (Hierarchical Mixture of Experts), to text categorization, instead of a single BP. They compared HME of BPs with a flat combination of BPs, and observed that HME is the better combination of BPs. Since BP learns training examples very slowly, it is not practical, in spite of its broad applicability and high accuracy, for implementing a text categorization system where training time is critical.

Research on machine learning based approaches to text categorization has been progressed very much, and they have been surveyed and evaluated systematically. In 1999, neural networks may be considered as an approach to text categorization, and among them, the MLP with BP is the most popular model [28].

The neural network model was initially created in 1986 by Mcelland and Rumelhart, and it was intended to performing tasks of pattern classification and nonlinear regressions as a supervised learning algorithm [29]. It was initially applied to text categorization in 1995 by Wiener [25]. Its performance was validated by comparing it with KNN in his master thesis on the test bed, Reuter21578. Even if the neural network classifies documents more accurately, it takes very much time for learning training documents.

The string kernel was proposed as the solution to the two main problems which is inherent in encoding documents into numerical vectors. It was initially proposed by Lodhi et al in 2002 as the kernel function of SVM [31]. String kernel receives two raw texts as its inputs and computes their syntactical similarity between them. Since documents don't need to be encoded into numerical vectors, the two main problems are naturally avoided. However, it costed very time for computing the similarity and failed to improve the performance of text categorization.

This research has three advantages as mentioned in this section. The first advantage of this research is to avoid the two main problems by encoding the documents into alternative structured data to numerical vectors. The second advantage is that string vectors are more transparent than numerical vectors with respect to the content of its full text; it is easier to guess the content of document by seeing its string vector than by its numerical vector specially when we want to classify some documents such Persian documents (Persian data are more complex). The third advantage as one derived from the second advantage is that it is potentially easier to trace why each document is classified. Therefore, this research proposes the novel method that creates a competitive self organizing neural network which received string vectors of documents data as its input data because of the three advantages.

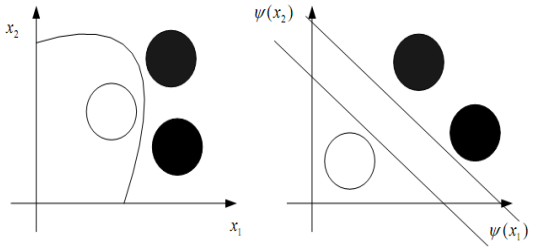

Figure 1. Mapping vector space in SVM

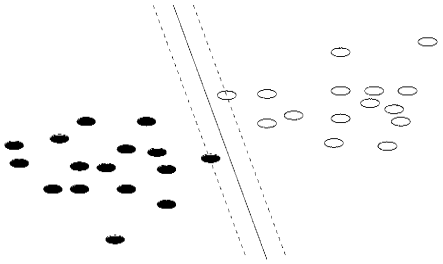

Figure 2. A decision line (solid) with a smaller margin

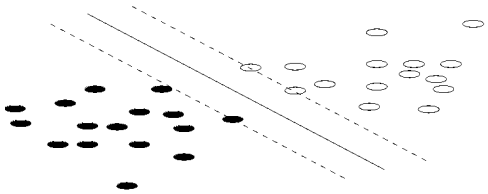

Figure 3. A decision line (solid) with the maximal margin

3. Strategies of Encoding Documents

Since the documents are unstructured data by themselves they cannot be processed directly by computers. They need to be encoded into structured data for processing them for text categorization. This section will describe the two strategies of encoding: the traditional strategy and the proposed strategy. The first subsection describes the formal description of TC, then the former, points out the two strategies of encoding documents.

3.1 Formal Description of TC Problem

Categorization is the task of assigning a Boolean value to each pair $< d_j, c_i > \in D \times C$ where D is a domain of documents and $C = \{c_1, c_2, ..., c_{|C|}\}$ is a set of predefined categories. A value of T assigned to $< d_j, c_i >$ indicates a decision to file d_j under c_i while a value of F indicates a decision not to file d_j under c_i. More formally the task is to approximate the unknown target function $\Phi := D \times C \rightarrow \{T, F\}$ (that describes how documents ought to be classified) by means of a function $\Phi := D \times C \rightarrow \{T, F\}$, called the classifier.

3.2 Numerical Vectors Vs String Vectors

A traditional strategy of encoding documents for tasks of text mining, such as text categorization is to represent them into numerical vectors. Since input vectors and weight vectors of traditional neural networks such as back propagation and RBF (Radial Basis Function) are given as numerical vectors, each document should be transformed into a numerical vector for using them for text categorization. Therefore, this subsection will describe the process of encoding documents into numerical vectors and what are their attributes and values.

Figure 4 illustrates the process of extracting feature candidates for numerical vectors from documents. If more than two documents are given as the input, all strings of documents are concatenated into a long string. The first step of this process is tokenization where the string is segmented into tokens by white space and punctuations. In the second step, each token is stemmed into its root form; for example, a verb in its past is transformed into its root form, and a noun in its plural form is transformed into its singular form. Words which function only grammatically with regardless of a content are called stop words [29], and they correspond to articles, conjunctions, or pronouns. In the third step, stop words are removed for processing documents more

efficiently and reliably for text categorization.

An alternative strategy of encoding documents for text categorization is to represent them into string vectors. In this part, we describe this strategy and its advantage in detail. However, this strategy is applicable to only proposed competitive self organizing neural network, while the previous one is applicable to any traditional machine learning algorithm.

A string vector is defined as a finite ordered set of words. In other words, a string vector is a vector whose elements are words, instead of numerical values. Note that a string vector is different from a bag of words, although both of them are similar as each other in their appearance. A bag of words is an infinite unordered set of words; the number of words is variable and they are independent of their positions. In string vectors, words are dependent on their positions as elements, since words correspond to their features.

Features of string vectors are defined as properties of words to the given document. The features are classified into the three types: linguistic features, statistical features, and positional features. Linguistic features are features defined based on linguistic knowledge about words in the given document: the first or last noun, verb, and adjective, in a paragraph, title, or full text. Statistical features are features defined based statistical properties of words in the given documents; the highest frequent word and the highest weighted word using following equation.

$$weight(w_k) = tf_i(wk)(\log_2 D - Log_2 df(w_k) + 1) \quad (1)$$

Where $tf_i(wk)$ is the frequency of words, w_k , D is the total number of document categories in corpus.

Positional features are features defined based on positions of words in a paragraph or the full text: a random word in the first or last sentence or paragraph, or the full text.

We can define features of string vectors by combining some of the three types, such as the first noun in the first sentence, the highest frequent noun in the first paragraph, and so on. A formal description of string vector is defined as a set of words which is ordered and has its fixed size. It is denoted by $[s_1, s_2, ..., s_d]$ where s_i denotes a string, and there are d elements. When representing documents into string vectors, their sizes are fixed with d, and it is called the dimension of string vectors. Since the elements are ordered in each string vector, two string vectors with their identical elements but different orders are treated as different ones. The reason is that each position of an element has its own different feature.

Table 1 illustrate differences between string vectors and numerical vectors. The first difference is that numerical values are given as elements in numerical vectors, while strings are given as elements in string vectors. The second difference is that the similarity measure between two numerical vectors is the cosine similarity or the Euclidean distance, while that between two string vectors is the semantic average similarity. The third difference between the two types of structured data is that features for encoding documents into numerical vectors are words, while those for encoding them into string vectors are statistical linguistic and posting properties of words. Therefore, a string vector is the vector where numerical values are replaced by strings in a numerical vector.

The differences between string vectors and bags of words are illustrated in table 2. Both types of structured data have strings as their elements.

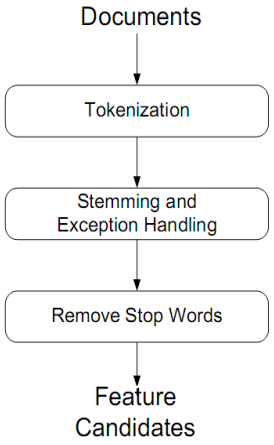

Figure 4. Flowchart of feature extraction of documents

As the similarity measure, cardinality of intersection of two bags of words is used while the average semantic similarity is used in string vectors. A bag of words is defined as an unordered infinite set of words, while a string vector is defined as an ordered finite set of words. Although a bag of words and a string vector look similar as each other, they should be distinguished from each other, based on table 2.

Table 1. The comparison of numerical and string vectors

	Numerical Vector	**String Vector**
Element	Numerical value	String
Similarity Measure	Inner products, Euclidean distance	Semantics similarity
Attributes	Words	Property of words

There are three advantages in representing documents into string vectors. The first advantage is to avoid completely the two main problems: the huge dimensionality and the sparse distribution. The second advantage is that the string vectors are characterized as more transparent representations of documents than numerical vectors; it is easier to guess the content of documents only from their representations. The third advantage is that there is the potential possibility of tracing more easily why each documents are classified so.

Figure 5 illustrates the process of encoding a document into its string vector with the simple definition of features. In the first step of figure 5, a document is indexed into a list of words and their frequencies. Its detail process of the first step is illustrated in figure 4. If the dimension of string vectors is set to d, d highest frequent words are selected from the list, in the second step. In the third step, the selected words are sorted in the descending order of their frequencies. This ordered list of words becomes a string vector representing the document given as the input.

This section describes the proposed competitive self organizing neural network, in detail, with respect to its architecture, training, classification, and properties.

Competitive self organizing neural networks belong to a class of recurrent networks, and-they are based on algorithms of unsupervised learning, such as the competitive algorithm explained in this section.

Table 2. The comparison of bag of words and string vectors

	Numerical Vector	**String Vector**
Element	String	
Similarity measure	Number of shared words	Semantics similarity
Set	Unordered infinite set	Ordered finite set

In competitive learning, the output neurons of a neural network compete among themselves to become active (to be "fired"). Whereas in MLP several output neurons may be active simultaneously, in competitive learning only a single output neuron is active at any time.

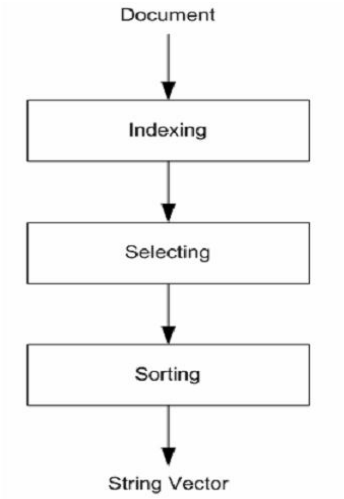

Figure 5. The process of mapping documents into a string vector

In other words, competitive learning is a learning procedure that divides a set of input patterns in clusters that are inherent to the input data. A competitive learning network is provided only with input vectors *x* and thus implements an unsupervised learning procedure.

A simple competitive learning network was depicted in Figure 6. A basic competitive network has an input layer and a competitive layer. The nodes in the competitive layer "compete" with each other, and the node that has the largest output becomes the "winning" neuron. The winning neuron is set to 1 and all other neurons are set to 0.

The training of the basic competitive network uses the Kohonen learning rule. For each input pattern, the weight vector of the winning node is moved closer to the input vector using the following formula:

$$w_i(q) = w_i(q-1) + \alpha(p(q) - w_i(q-1)) \quad (2)$$

Where w_i is the weight of the winning neuron, p is the corresponding input vector (string value) and D is the Kohonen learning rate. However, a problem of this model is that if the initial weight of a neuron is far from any vector, it will never be trained, so a bias vector is added to the result of the competition. The winning node would cause the bias vector to decrease. Under this mechanism, it is more difficult for a neuron to continue to win. The degree of bias is represented by a factor called conscience rate. As we show in figure 6 each of the four outputs *O* is connected to all inputs *i* with weight w_{i0}. When an input string vector *x* is presented only a single output unit of the network (the winner) will be activated. In a correctly trained network, all *x* in one cluster will have the same winner. For the determination of the winner and the corresponding learning rule, two methods exist: dot products and Euclidean distance. For simplicity of calculation we used the Euclidean distance in proposed network.

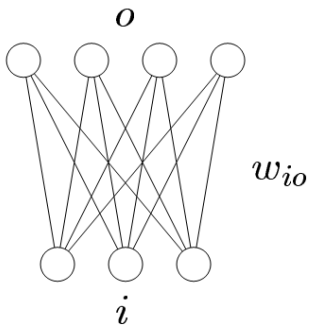

Figure 6. A simple competitive learning network

The proposed neural network follows self organizing map (SOM) in that synaptic weights are connected directly between the input layer and the competitive layer, and the weights are updated only when each training example is misclassified.

However, note that the proposed neural network is different from SOM in context of its detail process of learning and classification, since it uses string vectors as its input vectors, instead of numerical vectors. The competitive layer given as an additional layer to the input layer is different from the hidden layer of back propagation with respect to its role. The learning layer determines synaptic weights between the input and the competitive layer by referring to the tables owned by learning nodes. The learning of proposed neural network refers to the process of competition between weights stored in the tables.

Each training example is classified by summing the initial weights and selecting the category corresponding to the maximal sum. If the training example is classified correctly, the weights are not updated. Otherwise, the weights are incremented toward the target category and those are decremented toward the classified category. The winner weights (target category) are generated as the output of this process.

In the competitive neural network, each example is classified by summing the winner optimized weights, whether it is a training or unseen example. In addition weights connected to itself from the input nodes as its categorical score. The weights are decided by referring the table which is owned by its corresponding learning node. The category corresponding to the output node which generate its maximum categorical score (winner category) is decided as the category of the given example. Therefore, the output of this process is one of the predefined categories, assuming that the competitive neural network is applied to text categorization without the

decomposition. Figure 7 shows the diagram of proposed neural network. Complete algorithm of competitive neural text classifier and competitive learning algorithm was mentioned in classifier training algorithm and learning algorithm, respectively.

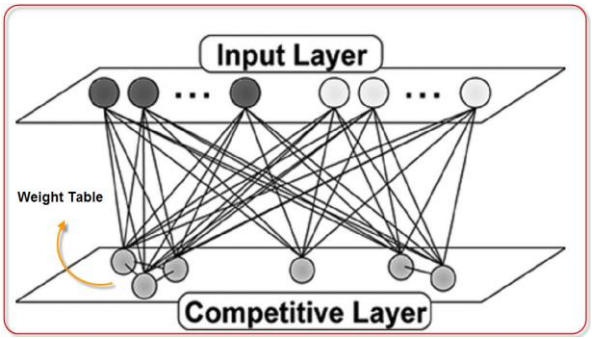

Figure 7. Proposed self organizing neural text classifier

Algorithm 1: Classifier Training

Input: A Series of Documents, Number of Categories
Output: The Winner Categories

1: Encode these sample documents into string vectors
2: Design the architecture of competitive text categorizer
3: Initialize weights in each learning node in competitive layer with its document
4: Repeat step 1-3 with the number of given documents
5: For each encoded sample document
6: Compute the values of winner nodes of the encoded
6-1: Classify each training vector into the corresponding category
6-2: Output: winner node in each learning node
6-2-1: If the winner node classify the documents correctly go to step 7
6-2-2: Update table weights
7: Output calculated weights
8: End

Algorithm 2: Winner Selection

Input: The Architecture of competitive self organizing neural text categorizer
Output: Selected Winner

1: Given one of string vector that in previous stage was created.
2: Compute the output value of nodes in the encoded document using the equation (2).
3: Classify the unseen string vector into the category corresponding to the output node (winner node)
4: End

4. Experimental Consideration

This section is concerned with the empirical validation of the performance of proposed method in several experiments.
An important issue of text categorization is how to measure the performance of the classifiers. Many measures have been used, each of which has been designed to evaluate some aspect of the categorization performance of a system [28]. In this section we discuss and analysis the important measures that have been reported in the literature.
We use the collection of Persian news categories, called irna.ir. In addition, For evaluating our method on English documents the standard test bed, Reuter 21578, was used. The Reuter 21578 is popularly used as the standard test bed for evaluating approaches to text categorization.
This set of experiments involves the five approaches: KNN, NB, SVM, NNBP, and our proposed method. In experiment result, the test bed and configurations of the approaches involved in the set of experiments are described, and the results of the set of experiments are presented and discussed.
The partition of the test bed, Reuter 21578 and irna.ir into the training and test set is illustrated in table 3 and 4, respectively. The test bed contains the most frequent categories of different type of news for entering the first evaluation, and its source is the web site, www.irna.ir. The collection was built by copying and pasting the news documents individually as the plain text files. In the test bed, the five categories and the 5,436 Persian and English news documents are available. The collection of news articles is partitioned into the training and test set by the ratio 7:3, as shown in table 3 and 4.

Table 3. Collection of different news articles on Reuter 21578

Category Name	Training Set	Test Set	Total
Trade	869	380	575
Earn	500	214	515
Grain	220	94	245
Wheat	430	185	615
Ship	250	110	360
Corn	280	120	400
Total	1890	820	2710

The configurations of the involved approaches are illustrated in table 5. The parameters of the SVM and the KNN, the capacity and the number of nearest neighbors, are set as five and six, respectively, but the NB has no parameter. The parameters of the NNBP such as the number of hidden nodes And the learning rate are arbitrary set as shown in table 5.
Persian news documents are encoded into 420 dimensional numerical vectors and 123 dimensional string vectors. English documents are encoded into 398 numerical vectors and 116 dimensional string vectors. We compared performance of the proposed method with four traditional approaches in following experiments.

4.1 Micro and Macro Averaging

For evaluating performance average across categories, there are two conventional methods, namely macro-averaging and micro-averaging. Macro-averaged performance scores are determined by first computing the performance measures per category and then averaging these to compute the global means. Micro-average performance scores are determined by first computing the totals of *a*, *b*, *c*, and *d* for all categories

and then use these totals to compute the performance measures. There is an important distinction between the two types of averaging. Micro-averaging gives equal weight to every document, while macro-averaging gives equal weight to each category. For evaluating the performance of the classifiers, we define four parameters:

• *a* - The number of documents correctly assigned to this category.
• *b* -The number of documents incorrectly assigned to this category.
• *c* - The number of documents incorrectly rejected from this category.
• *d* - The number of documents correctly rejected from this category.

The results of this experiment on Reuter 21578 test bed are presented in figure 8. Among the five methods, the left picture indicates the micro-averaged measure of each method. The right picture indicates the macro-averaged measure of each method, respectively. Our proposed approach shows its best performance to the NNBP, but the performance of our proposed approach is comparable to that of NNBP.

Table 4. Collection of different news articles on irna.ir

Category Name	Training Set	Test Set	Total
Politics	350	175	525
Law	360	145	505
Computer	150	75	225
Education	110	47	157
Economics	472	203	675
Sports	466	200	666
Total	1908	845	2753

Let's discuss the results from the set of experiments which were illustrated in figure 8. Even if the macro-averaged proposed neural network is not better than NNBP in the task, both are comparable to each other with respect to the performance of text categorization. Note that it requires very much time for training NNBP as the payment for the best performance. In addition, the NNBP is not practical in dynamic environments where NNBP must be trained again, very frequently. Hence, the proposed method is more recommendable than NNBP with respect to both the learning time and the performance.

4.2 F-Measure

Another evaluation criterion that combines recall and precision is the F-measure. In fact, the F1 measure is used for evaluating the performance of TC. The F1measure can be calculated as following equation:

$$E(P) = \sum \frac{(1+\beta) \times \mathrm{Re}call\,(i,k) \times precision(i,k)}{\beta \times \mathrm{Re}call\,(i,k) + precision(i,k)} \quad (3)$$

Precision and recall are widely used for evaluation measures in TC. For calculating the F1 measure, in each category and each documents we should determines whether the document belongs to the category or not. So we need to define the recall and precision rate with the parameters that defined in previous section as:

$$recall = \frac{a}{a+c}$$
$$precision = \frac{a}{a+b} \quad (4)$$

Table 5. Parameter settings of algorithms

Algorithms	Parameter Settings
SVM	Capacity = 5.0
KNN	# Nearest Value= 6
Naïve Bayes	N/A
NN With Back Propagation (BP)	# Hidden Layer=15 Learning Rate=0.2 #Training Epoch=500
Proposed Method	Learning Rate=0.2 #Training Epoch=150

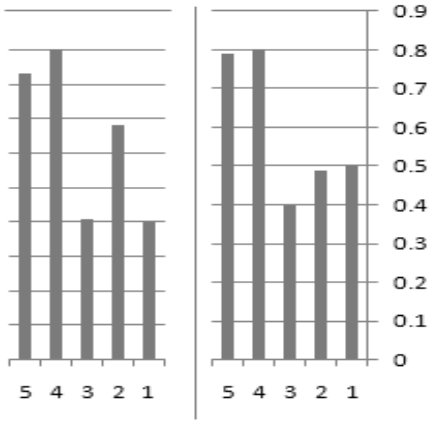

Figure 8. The left side of figure indicates the micro-averaged measure and the right one shows the macro-averaged for (left to right): proposed method, NNBP, SVM, NB, KNN

Figure 9 shows the result of evaluating the F1 measure for five approaches on the irna.ir test bed. Science each category contain identical number of test documents, micro-averaged and macro-averaged F1 are same as each other. Therefore, their performances are presented in an integrated group, instead of two separated groups, in figure 9. This result shows that back propagation is the best approach in comparison to another three traditional algorithms, while NB is the worst approach with the decomposition of the task on this test bed. Unlike the previous experiment set, NTC is comparable and competitive with back propagation. So we discuss this analysis in next subsections with combined to another experiments

4.3 Accuracy

Figure 10 show the accuracy of all methods on Reuter 21578 news document test bed. This picture show that the proposed neural network has more reliable than other traditional method.

The accuracy rate of the proposed neural network on test bed is more than 86% but the best traditional approach have 80% accuracy rate.

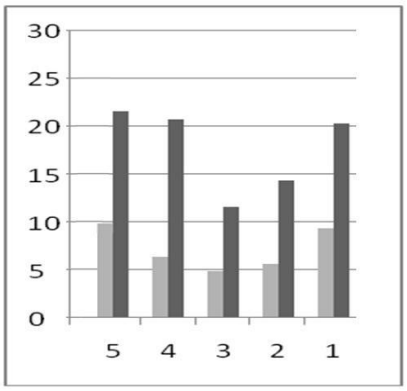

Figure 9. The F1 measure evaluation for (left to right): proposed method, KNN, NB, SVM, NNBP

4.4 Recall and Precision Rate

We also tried another performance measure for our proposed method to show the quality of document classification. We validate the performance of novel approach by comparing it with other machine learning algorithms on the irna.ir test bed in this experiment. Table 7 shows these rate for best traditional method and novel method.

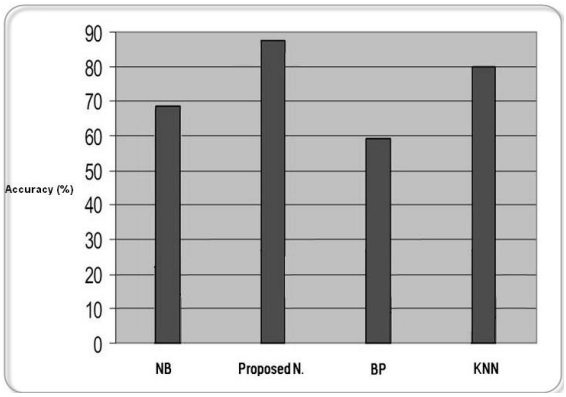

Figure 10. Accuracy rate of news classifier on Reuter 21578

From the above result (and section 4.2) we can see that the documents classifier based on competitive self organizing neural network with string vector can classify document of different categories correctly, represented by a high accuracy rate. If we use the positive and negative accuracy rate for evaluating the performance of proposed text classifier we have:

Table 6. Positive and negative accuracy for news text classifier

Positive Accuracy	Negative Accuracy	Average Accuracy
0.4894	0.9368	0.7131

Where a low positive accuracy rate shows that many documents from different categories are not clustered together. This is partly because it is difficult to have information about the correct category in unsupervised learning. In addition this result shows that the classifiers based on BP network (NNBP) couldn't classify documents (in most categories) correctly.

Table 7. Precision and Recall rate of best traditional and novel text classifier

	Precision	Recall
SVM	0.6398	0.4
NNBP	0.4367	0.4
KNN	0.5612	0.8
NB	0.7866	0.65
Our Method	0.9107	0.9

In SVM experiment, precision and recall are low in some categories. The novel method takes less than one-tenth of the time BP takes when training. At the same time, it performs well in the categories in which the results are satisfactory. In the different news category, it even outperforms the NB method. Figure 11 shows the complete recall and precision rate on some category on irna.ir test bed. This picture shows the robustness and quality of text categorization by the competitive self organizing neural text categorizer. The novel method can over perform the traditional method with classify precision rate of 0.8.

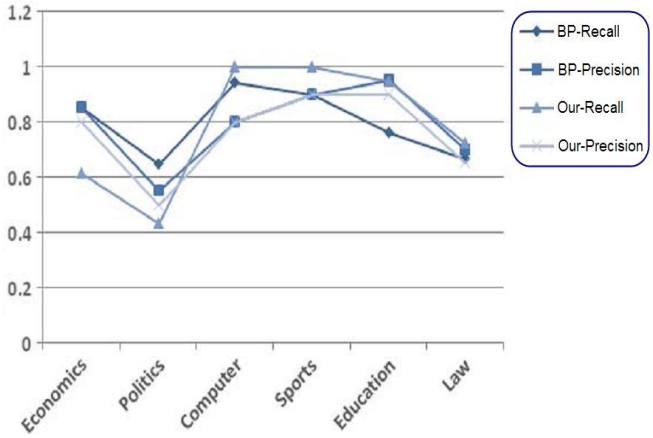

Figure 11. Precision and Recall rate between best traditional and novel algorithm on each categories

5. Conclusion

This research proposes a novel method that used competitive self organizing neural network with string vector for text categorization which uses alternative representations of documents to numerical vectors. In this method we used a full inverted index as the basis for the operation on string vectors, instead of a restricted sized similarity matrix. It was cheaper to build an inverted index from a corpus than a similarity matrix, as mentioned in section 2. In the previous attempt, a restricted sized similarity matrix was used as the basis for the operation on string vectors. Therefore, information loss from the similarity matrix degraded the performance of the modified version. This research addresses the information loss by using a full inverted index, instead of a restricted sized similarity matrix.

The four contributions are considered as the significance of this research. For first, this research proposes the practical approach for documents categorization, according to the results of the set of experiments. For second, it solved the two main problems, the huge dimensionality and the sparse

distribution which are inherent in encoding documents into numerical vectors. For third, it created a new neural network, called competitive self organizing neural text categorizer, which receives string vectors differently from the previous neural networks. For last, it provides the potential easiness for tracing why each news document is classified so. Other machine learning algorithms such as Naïve Bayes and back propagation are considered to be modified into their adaptable versions to string vectors. The operation may be insufficient for modifying other machine learning algorithms. For example, it requires the definition of a string vector which is representative of string vectors corresponding to a mean vector in numerical vectors for modifying k-means algorithm into the adaptable version. Various operations on string vectors should be defined in a future research for modifying other machine learning algorithms.

Let's consider another remaining task as the further research. The first task is to apply the proposed competitive self organizing neural network to categorization of documents within a specific domain such as medicine, law, and engineering. The second task is to modify it into the static version.

References

[1] K. Androutsopoulos, K. V. Koutsias, Chandrinos, C. D. Spyropoulos, "An Experimental Comparison of Naïve Bayes and Keyword-based Anti-spam Filtering with Personal Email Message", Proceedings of 23rd ACM SIGIR, pp.160-167, 2000.

[2] N. Cristianini, J. Shawe-Taylor, "Support Vector Machines and Other Kernel-based Learning Methods", Cambridge University Press, 2000.

[3] N.L. Bhamidipati, S.K. Pal, "Stemming via Distribution-Based Word Segregation for Classification and Retrieval", IEEE Transactions on Systems Man. and Cybernetics, Vol.37, No.2, pp.350-360, 2007.

[4] R.O. Duda, P.E. Hart, D.G. Stork, "Pattern Classification", John Wiley & Sons, Inc, 2001.

[5] V.I. Frants, J. Shapiro, V.G. Voiskunskii, "Automated Information Retrieval: Theory and Methods", Academic Press, 1997.

[6] M.T. Hagan, H.B. Demuth, M. Beale, "Neural Network Design", PWS Publishing Company, 1995.

[7] S. Haykin, "Neural Networks: Comprehensive Foundation", Macmillan College Publishing Company, 1994.

[8] A. Frolov, D. Husek, "Recurrent Neural Network based Boolean Factor Analysis and Its Application to Word Clustering", IEEE Transactions on Neural Networks, Vol.20, No.7, pp.1073-1086, 2009.

[9] P. Jackson, I. Mouliner, "Natural Language Processing for Online Applications: Text Retrieval, Extraction and Categorization", John Benjamins Publishing Company, 2002.

[10] T. Martin, H.B. Hagan, H. Demuth, M. Beale, "Neural Network Design", PWS Publishing Company, 1995.

[11] L. Man, S. Jian, "Empirical Investigations into Full-Text Protein Interaction Article Categorization Task (ACT) in the Procreative II Challenge", IEEE/ACM Transactions on Computational Biology and Bioinformatics, Vol.7, No.3, pp.421-427, 2010.

[12] B. Massand, G. Linoff, D. Waltz, "Classifying News Stories using Memory based Reasoning", Proceedings of 15th ACM International Conference on Research and Development in Information Retrieval, pp.59-65, 1992.

[13] Y. Yang, "An evaluation of statistical approaches to text categorization", Information Retrieval, Vol.1, No.1-2, pp.67-88, 1999.

[14] F. Sebastiani, "Machine Learning in Automated Text Categorization", ACM Computing Survey, Vol.34, No.1, pp.1-47, 2002.

[15] J. Rennie, "Improving Multi-class Text Classification with Support Vector Machine", Master's thesis, Massachusetts Institute of Technology, 2001.

[16] T. M. Mitchell, "Machine Learning", McGrawHill, 1997.

[17] D. Mladenic, M. Grobelink, "Feature Selection for Unbalanced Class Distribution and Naïve Bayes", Proceedings of International Conference on Machine Learning, pp.256-267, 1999.

[18] M.E. Ruiz, P. Srinivasan, "Hierarchical Text Categorization Using Neural Networks", Information Retrieval, Vol.5, No.1, pp.87-118, 2002.

[19] J.C. Platt, "Sequential Minimal Optimization: A Fast Algorithm for Training Support Vector Machines", Technical Report MSR-TR-98-14, 1998.

[20] M. Hearst, "Support Vector Machines", IEEE Intelligent Systems, Vol.13, No.4, pp.18-28, 1998.

[21] T. Joachims, "Text Categorization with Support Vector Machines: Learning with many Relevant Features", The Proceedings of 10th European Conference on Machine Learning, pp.143-151, 1998.

[22] D.A. Bell, J.W. Guan, "On Combining Classifier Mass Functions for Text Categorization", IEEE Transactions on Knowledge and Data Engineering, Vol.17, No.10, pp.1307-1319, 2005.

[23] H. Drucker, D. Wu, V.N. Vapnik, "Support Vector Machines for Spam Categorization", IEEE Transaction on Neural Networks, Vol.10, No.5, pp.1048-1054, 1999.

[24] P.G Espejo, S. Ventura, "A Survey on the Application of Genetic Programming to Classification", Systems, IEEE Transactions on Man. and Cybernetics, Vol.40, No.2, pp.121-144, 2010.

[25] E.D. Wiener, "A Neural Network Approach to Topic Spotting in Text", The Thesis of Master of University of Colorado, 1995.

[26] http://www.research.att.com/~lewis/reuters21578.html.

[27] F. Gabriel Pui Cheong, J. Yu, "Text Classification without Negative Examples Revisit", IEEE Transactions on Knowledge and Data Engineering, Vol.18, No.1, pp.6-20, 2006.

[28] D. Isa, L. Lee, "Text Document Preprocessing with the Bayes Formula for Classification Using the Support Vector Machine", IEEE Transactions on Knowledge and Data Engineering, Vol.20, No.9 pp.1264-1272, 2008.

[29] V. Lertnattee, T. Theeramunkong, "Multidimensional Text Classification for Drug Information", IEEE Transactions on Information Technology in Bio medicine, Vol.8, No.3, pp.306-312, 2008.

[30] L. Man, Chew, T. Lim, "Supervised and Traditional Term Weighting Methods for Automatic Text Categorization", IEEE Transactions on Pattern Analysis and Machine Intelligence, Vol.31, No.4, pp.721-735, 2009.

[31] H. Lodhi, C. Saunders, J. Shawe-Taylor, N. Cristianini, C. Watkins, "Text Classification with String Kernels", Journal of Machine Learning Research, Vol.2, No.2, pp.419-444, 2002.

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Separation of Tabla from Singing Voice using Percussive Feature Detection in a Polyphonic Channel

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Abstract - *In many signal processing applications, different sources of sound are to be separated for further modifications. Here, a method for the separation of tabla sound from a mixer of vocal and tabla is presented. For this, the short-time Fourier transform (STFT) of the mixed signal is taken. The log difference of each frequency component between consecutive frames in the magnitude spectra is obtained. If the log difference of the magnitude of frequency components exceeds a user specified threshold value (Th) the bin corresponding to that position is incremented by '1'. If the threshold condition is met, it is deemed to belong to a percussive onset. The final value of this counter, once each frequency bin has been analyzed, is then taken to be a measure of percussivity of the current frame. Once all frames have been processed, we have a temporal profile which describes the percussion characteristics of the signal. This profile is then used to modulate the spectrogram before resynthesis. The magnitude of all the frequency components in the consecutive frames for which the log difference exceeds threshold value is then added with the original phase. The inverse FFT is then computed of the resultant signal to generate a signal which is none other than separated tabla signal. In addition to producing high quality separation results, the method we describe is also a useful pre-process for tabla transcription in the mixer of tabla and vocal. Although the separated tabla sound does not contain any residual of vocal sound, the quality of the sound needs to be further enhanced. The quality of the o/p signal is further smoothening by using a straight method which is a very popular technique for signal analysis and synthesis.*

1. Introduction

Only a few systems directly address the separation of music instrument from singing voice. A system proposed by meron and hirose [1] aims to separate piano accompaniment from singing voice In recent years, some focus has shifted from pitched instrument transcription to drum transcription [2]; and likewise in the field of sound source separation, some particular attention has been given to drum separation in the presence of pitched instruments [3]. Algorithms such as ADRess [4] and those described in [5] are capable of drum separation in stereo signals if certain constraints are met. Other algorithms such as [6] [7] have attempted drum separation from single polyphonic mixture signals with varying results. In this paper we present a fast and efficient way to decompose a spectrogram using a simple technique which involves percussive feature detection which results in the extraction of the tabla parts from a polyphonic mixture

of singing voice. The quality of the o/p signal is further smoothened by using a straight method which is a very popular technique for signal analysis and synthesis. The algorithm is applicable for the separation of almost any audio features which exhibit rapid broadband fluctuations such as tabla in music or plosives, fricatives and transients in speech. Automatic tabla separation and transcription is in itself can be a useful tool in applications such as processing of old song records. In the present work we investigate audio features suitable for use in a threshold based detector to detect tabla segments from a mixture of tabla and singing voice.

2. Method Overview

Tabla used in popular music can be characterized by a rapid broadband rise in energy followed by a fast decay. The tabla consists of a pair of drums, one large base drum, the bayan, and a smaller treble drum, the dayan. Tabla percussion consists of a variety of strokes, often played in rapid succession, each labeled with a mnemonic. Two broad classes of strokes, in terms of acoustic characteristics, are: 1. tonal strokes that decay slowly and have a near-harmonic spectral structure and 2. impulsive strokes that decay rapidly and have a noisy spectral structure. The pitch percept by tonal tabla strokes falls within the pitch range of the human singing voice. It was found that while all the impulsive strokes had similar acoustic characteristics, there was a large variability in those of the different tonal strokes. On the other hand the pitch dynamics (the evolution of pitch in time) of singing voice tends to be piece-wise constant with abrupt pitch changes in between. A percussive temporal profile is derived by computing STFT of the signal per frame and assigning a percussive measure to it. The frame is then scaled according to this measure. It should be seen that regions of the spectrogram with low percussive measures will be scaled down significantly. Upon resynthesis, only the percussive regions remain.

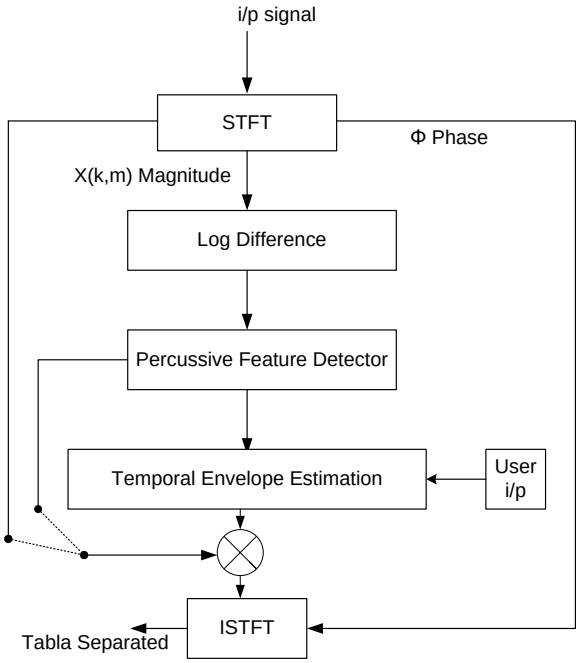

Figure 1. System Overview

The above Figure1, demonstrates the general operation of the algorithm. The magnitude STFT of the mixture (Tabla+Vocal) is taken, while the phase (Φ) is retained for resynthesis purposes at the later stage. The log difference of each frequency component between consecutive frames in the magnitude spectra is obtained. This measure effectively tells us how rapidly the spectrogram is fluctuating. If the log difference of

the magnitude of frequency components exceeds a user specified threshold value (T_h) the bin corresponding to that position is incremented by '1'. If the threshold condition is met, it is deemed to belong to a percussive onset and a counter is incremented. The final value of this counter, once each frequency bin has been analyzed, is then taken to be a measure of percussivity of the current frame. Once all frames have been processed, we have a temporal profile which describes the percussion characteristics of the signal. This profile is then used to modulate the spectrogram before resynthesis. The magnitude of all the frequency components in the consecutive frames for which the log difference exceeds threshold value is then added with the original phase. The inverse FFT is then computed of the resultant signal to generate a signal which is none other than separated tabla signal. It is found that separated o/p contains little bit of noise which is further smoothened by using Straight Algorithm method for signal analysis and synthesis.

3. System Implementation

The given i/p signal comprises of a mixture of vocal & tabla signal. The short-time Fourier transform (STFT) of the mixed signal is taken. The magnitude X(k,m) STFT (Short Time Fourier Transform) of the given input signal is taken and the phase(Φ) is retained for resynthesis purposes later on.

$$X(k,m) = abs\left[\sum_{n=0}^{N-1} w(n).x(n+mH).e^{-j2\pi nk/N}\right] \text{--} \mathbf{1}$$

where X(k,m) is the absolute value of the complex STFT given in equation 1 and where m is the time frame index, k is the frequency bin index, H is the hop size between frames and N is the FFT window size and where w(n) is a suitable window of length N also.

The log difference of each frequency component between consecutive frames in the magnitude spectra is obtained.

$$X'(k,m) = 20\log_{10}\frac{X(k,m-1)}{X(k,m)} \quad \text{---}\mathbf{2}$$

For all m and $1 \le k \le K$

Where $X'(k,m)$ is the log difference of the spectrogram w. r. t. time.

If the log difference of the magnitude of frequency components exceeds a user specified threshold value (T_h) the bin corresponding to that position is incremented by '1'. If the threshold condition is met, it is deemed to belong to a percussive onset and a counter is incremented.

$$Pe(m) = \sum_{k=1}^{K}\begin{cases} P(k,m)=1 & if \quad X'(k,m)>T \\ P(k,m)=0 & otherwise \end{cases} \text{ -- } \mathbf{3}$$

Where *Pe*(m) is the percussive measure and *T* is the threshold value. The Final value of this counter once each frequency bin has been analyzed is then taken to be a measure of percussivity of the current frame. Once all frames have been processed, we have a temporal profile which describes the percussion characteristics of the signal. This profile is then used to modulate the spectrogram before resynthesis.

$$Y(k,m) = Pe(m)^{\psi} X(k,m) \text{ -- } \mathbf{4}$$

For all m and $1 \le k \le K$

The magnitude of all the frequency components in the consecutive frames for which the log difference exceeds threshold value is then added with the original phase (Φ).

$$Y(k,m) = Pe(m)^{\Psi}.X(k,m).P(k,m) \text{ -- } \mathbf{5}$$

The inverse FFT is then computed of the resultant signal to generate a signal which is none other than separated Tabla signal.

$$Y(n+mH) = w(n)\left[\frac{1}{K}\sum_{k=1}^{K} Y(k,m).e^{j\angle x_{\omega}(k,m)}\right]^{norm} \text{ -- } \mathbf{6}$$

4. Results

The investigations were carried out using different proportions of tabla and vocal sounds. It has been observed from the various experiments that by keeping the proportion of tabla fixed and by varying the proportion of vocal in a given mixture; we can achieve the separation of tabla from the given mixture. But, it has also been observed that when the vocal is at very less proportion in a given mixture, we get a very pure tabla signal in the o/p signal. As we go on increasing the proportion of vocal in a given mixture, the quality of separated tabla signal is degraded. It has been seen that when the vocal proportion is 50% of tabla in a given mixture, we get a distorted tabla signal in the o/p. when we kept the threshold value (T_h) at –ve level; we get a mixed (Tabla + Vocal) signal in the o/p. The desired o/p results are obtained by keeping the value of threshold parameter at fixed level and by varying the value of chip (Ψ- power). Therefore, we can say that if the proportion of vocal is very less, i.e. 0.01 of tabla in a given mixture, the value of chip (Ψ- power) will keep at high value, on the other hand, if the proportion of vocal is 0.5 of tabla in a given mixture, the value of chip (Ψ- power) needs to be kept at low.

The analysis of the results is carried out using time domain waveforms & spectrograms. The smoothness of the time domain waveforms indicates the good quality of the separated out tabla sound. The spectrograms represent the frequency contents of the signal with respect to time. Smooth transition of the spectrogram segments represent good quality of the separated out tabla sound.

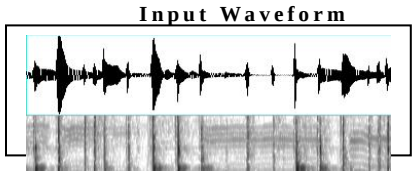

Figure 2. Original waveform & spectrogram of mixed Tabla + Vocal01.

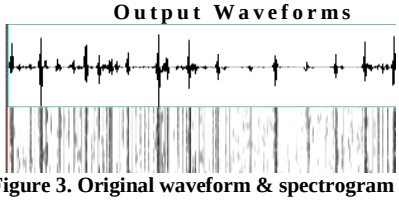

Figure 3. Original waveform & spectrogram of mixed Tabla + Vocal01_pm using Percussive method.

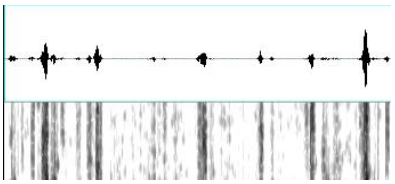

Figure 4. Original waveform & spectrogram of mixed Tabla + Vocal01_pm_st using Straight method.

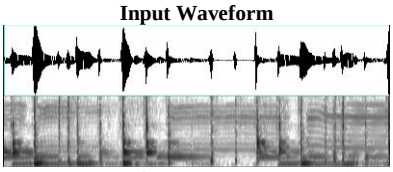

Figure 5. Original waveform & spectrogram of mixed Tabla + Vocal02.

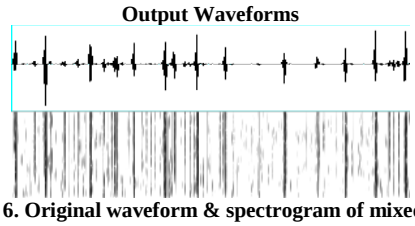

Figure 6. Original waveform & spectrogram of mixed Tabla + Vocal02_pm using Percussive method.

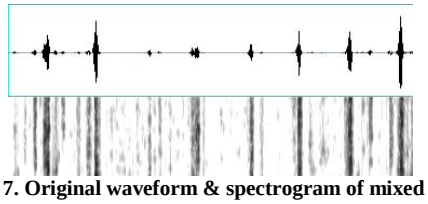

Figure 7. Original waveform & spectrogram of mixed Tabla + Vocal02_pm_st using Straight method.

5. Conclusions

In this thesis investigation were carried out to evaluate the quality of the separated out tabla sound from the mixture of different proportion of tabla and vocal sounds. Two techniques were used for separating out the tabla sound. In the first technique, the percussiveness of the tabla was exploited to separate the tabla sound. In the 2nd technique, the separate out tabla sounds from technique first was further smoothened by using the straight technique for improving the quality of the output. In addition to producing high quality separation results, the method we describe is also a useful pre-process for tabla transcription in the mixer of tabla and vocal. Although the separated tabla sound does not contain any residual of vocal sound, the quality of the sound needs to be further enhanced. It was also observed that there is a limit of the proportion of vocal sound for obtaining satisfactory quality of the tabla sound from the mixture. It means that if we increase the proportion of the vocal sound above this limit, the quality of the separated out tabla sound is deteriorated. If we go on raising the limit of vocal in the given mixture, the tabla sounds in the output gets distorted.

References

[1] Y Meron and K. Hirose, "Separation of singing and piano sounds," in Proceedings of the 5th International Conference on Spoken Language Processing, 1998.

[2] D. FitzGerald, E. Coyle, and B. Lawlor, "Sub-band independent subspace analysis for drum transcription," in Proceedings of the Digital Audio Effects Conference, Hamburg, pp. 65-69, 2002.

[3] D. FitzGerald, E.Coyle, and B. Lawlor, "Drum transcription in the presence of pitched instruments using prior subspace analysis," in Proceedings of the Irish Signals and Systems Conference 2003, Limerick. July 1-2 2003.

[4] D. Barry, R. Lawlor, and E. Coyle, "Real-time sound source separation using azimuth discrimination and resynthesis," in Proceedings of the Audio Engineering Society Convention, October 28-31, San Francisco, CA, USA, 2004.

[5] C. Avendano, "Frequency domain source identification and manipulation in stereo mixes for enhancement, suppression and re-panning applications," in Proceedings of the IEEE

Workshop on Applications of Signal Processing to Audio and Acoustics, pp. 55-58 New Paltz, NY, October 19-22, 2003.

[6] A. Zils, F. Pachet, O. Delerue, and F. Gouyon, "Automatic extraction of drum tracks from polyphonic music signals," in Proceedings. of the 2nd International Conference on Web Delivering of Music, Darmstadt, Germany, Dec. 9-11, 2002.

[7] C. Uhle, C. Dittmar, and T. Sporer, "Extraction of drum tracks from polyphonic music using independent subspace analysis," in Proceedings of 4th International Symposium on Independent Component Analysis and Blind Signal Separation, April 2003, Nara, Japan.

[8] D. Barry, R. Lawlor, and E. Coyle, "Comparison of signal reconstruction methods for the azimuth discrimination and resynthesis algorithm," in Proceedings of the 118th Audio Engineering Society Convention, May 28-31, Barcelona, Spain, 2005.

[9] D. Barry, R. Lawlor, and E. Coyle, "Drum Source Separation using Percussive Feature Detection and Spectral Modulation," in ISSC 2005,Dublin, September 1-2 2005.

[10] V. Rao and P. Rao, " Vocal melody detection in the presence of pitched accompaniment using harmonic matching methods", in Proceedings of the 11th International Conference on Digital Audio Effects (DAFx-08), Espoo, Finland, 2008.

[11] H. Kawahara, I. Masuda-Katsuse and A. de Cheveigne, "Restructuring speech representations using a pitch-adaptive time frequency smoothing and an instantaneous-frequency based F0 extraction", Speech Communication, vol. 27, pp. 187–207, 1999.

[12] A. Bapat, V. Rao, and P. Rao, "Melodic contour extraction of Indian classical vocal music", in Proceedings of International Workshop on Artificial Intelligence and Music (Music-AI '07), Hyderabad, Jan. 2007.

An Alternative Method of Finding the Membership of a Fuzzy Number

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Abstract: In this article, it has been shown that the Dubois-Prade left and right reference functions of a fuzzy number viewing as a distribution function and a complementary distribution function respectively, leads to a very simple alternative method of finding the membership of any function of a fuzzy number. This alternative method has been demonstrated with the help of different examples.

Key words and phases: Fuzzy membership function, Dubois-Prade reference function, Distribution function, Superimposition.

1. Introduction

Finding the fuzzy membership function (fmf) using the standard method of α-cuts is sometime impossible. For example, the method of α-cuts fails to find the fmf of even the simple function $\sqrt{X}$ when X is fuzzy. Indeed, for $\sqrt{X}$ in particular, Chou (2009) has actually forwarded a method of finding the fmf for triangular fuzzy number (tfn) X . We shall in this article, put forward an alternative method of finding the membership of functions of fuzzy numbers.

Dubois and Prade (Kaufmann and Gupta (1984)) have defined a fuzzy number $X = [a, b, c]$ with membership function

$$\mu_X(x) = \begin{cases} L(x), a \le x \le b \\ R(x), b \le x \le c \\ 0, \ otherwise \end{cases} \qquad (1.1)$$

$L(x)$ being a continuous non-decreasing function in the interval [a, b], and $R(x)$ being a continuous non-increasing function in the interval [b, c], with $L(a) = R(c) = 0$ and $L(b) = R(b) = 1$. Dubois and Prade named $L(x)$ as left reference function and $R(x)$ as right reference function of the concerned fuzzy number. A continuous non-decreasing function of this type is also called a distribution function with reference to a Lebesgue-Stieltjes measure (de Barra, pp-156).

In this article, we are going to show how easy the whole process of finding an fmf can be if start from the simple assumption that the Dubois-Prade left reference function is a distribution function in the measure theoretic sense, and similarly the Dubois-Prade right reference function is complementary distribution function. Accordingly, the functions $L(x)$ and $(1 - R(x))$ would have to be associated with densities $\frac{d}{dx}(L(x))$ and $\frac{d}{dx}(1 - R(x))$ in [a, b] and [b, c] respectively (Baruah (2010 a, b)).

2. Superimposition of Sets

The superimposition of sets defined by Baruah (1999) is defined as such that if the set A is superimposed over the set B, we get

$$A(S)\, B = (A-B) \cup (A \cap B)^{(2)} \cup (B-A) \qquad (2.1)$$

where S represents the operation of superimposition, and $(A \cap B)^{(2)}$ represents the elements of $(A \cap B)$ occurring twice. It can be seen that for two intervals $[a_1, b_1]$ and $[a_2, b_2]$ superimposed gives

$$[a_1, b_1]\,(S)\,[a_2, b_2]$$

$$= [a_{(1)}, a_{(2)}] \cup [a_{(2)}, b_{(1)}]^{(2)} \cup [b_{(1)}, b_{(2)}]$$

where $a_{(1)} = \min(a_1, a_2)$, $a_{(2)} = \max(a_1, a_2)$, $b_{(1)} = \min(b_1, b_2)$, and $b_{(2)} = \max(b_1, b_2)$.

Indeed, in the same way if $[a_1, b_1]^{(1/2)}$ and $[a_2, b_2]^{(1/2)}$ represent two uniformly fuzzy intervals both with membership value equal to half everywhere, superimposition of $[a_1, b_1]^{(1/2)}$ and $[a_2, b_2]^{(1/2)}$ would give rise to

$$[a_1, b_1]^{(1/2)}\,(S)\,[a_2, b_2]^{(1/2)}$$

$$= [a_{(1)}, a_{(2)}]^{(1/2)} \cup [a_{(2)}, b_{(1)}]^{(1)} \cup [b_{(1)}, b_{(2)}]^{(1/2)}. \qquad (2.2)$$

So for n fuzzy intervals $[a_1, b_1]^{(1/n)}$, $[a_2, b_2]^{(1/n)}$,…, $[a_n, b_n]^{(1/n)}$ all with membership value equal to1/n everywhere, $[a_1, b_1]^{(1/n)}$(S) $[a_2, b_2]^{(1/n)}$(S) …… (S) $[a_n, b_n]^{(1/n)}$

$$= [a_{(1)}, a_{(2)}]^{(1/n)} \cup [a_{(2)}, a_{(3)}]^{(2/n)} \cup \ldots\ldots\ldots \cup [a_{(n-1)}, a_{(n)}]^{((n-1)/n)} \cup [a_{(n)}, b_{(1)}]^{(1)} \cup [b_{(1)}, b_{(2)}]^{((n-1)/n)} \cup \ldots\ldots\ldots \cup [b_{(n-2)}, b_{(n-1)}]^{(2/n)} \cup [b_{(n-1)}, b_{(n)}]^{(1/n)}, \quad (2.3)$$

where, for example, $[b_{(1)}, b_{(2)}]^{((n-1)/n)}$ represents the uniformly fuzzy interval $[b_{(1)}, b_{(2)}]$ with membership ((n-1) /n) in the entire interval, $a_{(1)}, a_{(2)}, \ldots, a_{(n)}$ being values of $a_1, a_2, \ldots, a_n$ arranged in increasing order of magnitude, and $b_{(1)}, b_{(2)}, \ldots, b_{(n)}$ being values of $b_1, b_2, \ldots, b_n$ arranged in increasing order of magnitude.

We now define a *random* vector $X = (X_1, X_2, \ldots, X_n)$ as a family of X_k, k = 1, 2,…, n, with every X_k inducing a sub-σ field so that X is measurable. Let $(x_1, x_2 \ldots x_n)$ be a particular realization of X, and let $X_{(k)}$ realize the value $x_{(k)}$ where $x_{(1)}, x_{(2)}, \ldots\ldots, x_{(n)}$ are *ordered* values of $x_1, x_2, \ldots\ldots, x_n$ in increasing order of magnitude. Further let the sub-σ fields induced by X_k be independent and identical. Define now

$$\Phi_n(x) = 0, \text{ if } x < x_{(1)},$$

$$= (r-1)/n, \text{ if } x_{(r-1)} \leq x \leq x_{(r)}, r = 2, 3, \ldots, n,$$

$$=1, \text{ if } x \geq x_{(n)}; \quad (2.4)$$

$\Phi_n(x)$ here is an empirical distribution function of a theoretical distribution function Φ(x).

As there is a one to one correspondence between a Lebesgue-Stieltjes measure and the distribution function, we would have

$$\Pi(a, b) = \Phi(b) - \Phi(a) \quad (2.5)$$

where Π is a measure in (Ω, *A*, *Π*), *A* being the σ- field common to every x_k.

Now the Glivenko-Cantelli theorem (see e.g. Loeve (1977), pp-20) states that $\Phi_n(x)$ converges to Φ(x) uniformly in x. This means,

$$\sup \left| \Phi_n(x) - \Phi(x) \right| \to 0. \quad (2.6)$$

Observe that (r-1)/n in (2.4), for $x_{(r-1)} \leq x \leq x_{(r)}$, are membership values of $[a_{(r-1)}, a_{(r)}]^{((r-1)/n)}$ and $[b_{(n-r+1)}, b_{(n-r)}]^{((r-1)/n)}$ in (2.3), for r = 2, 3,…, n. Indeed this fact found from superimposition of uniformly fuzzy sets has led us to look into the possibility that there could possibly be a link between distribution functions and fuzzy membership.

3. The fmf of $\sqrt{X}$ for a tfn X

Consider a tfn $X = [a, b, c],\ (a,\ b,\ c > 0)$, with fmf

$$\mu_X(x) = \begin{cases} L(x) = \frac{x-a}{b-a}, a \leq x \leq b \\ R(x) = \frac{c-x}{c-b}, b \leq x \leq c \\ 0, \quad otherwise \end{cases} \quad (3.1)$$

where $L(x)$ is the left reference function and $R(x)$ is the right reference function. $L(x)$ is a distribution function and $R(x)$ is a complementary distribution function. Accordingly, the density functions for the distribution functions $L(x)$ and $(1 - R(x))$ be $f(x)$ and $g(x)$. Then,

$$f(x) = \frac{d}{dx}(L(x)) = \frac{1}{b-a}, \quad a \leq x \leq b$$

and $g(x) = \frac{d}{dx}(1 - R(x)) = \frac{1}{c-b}, \ b \leq x \leq c.$

Let $y = \sqrt{x}$ so that $dx/dy = 2y.$

The distribution function for, would now be given by

$$\int_{\sqrt{a}}^{x} \frac{1}{b-a}.2ydy = \frac{x^2-a}{b-a}, \ \sqrt{a} \leq x\sqrt{b},$$

and the complementary distribution for $\sqrt{X}$is

$$1 - \int_{\sqrt{b}}^{x} \frac{1}{c-b}.2ydy = 1 - \frac{x^2-b}{c-b}, \ \sqrt{b} \leq x\sqrt{c}.$$

Indeed this distribution function and the complementary distribution function constitute the fmf of $\sqrt{X}$ as,

$$\mu_{\sqrt{X}}(x) = \begin{cases} \frac{x^2-a}{b-a}, \sqrt{a} \leq x \leq \sqrt{b} \\ \frac{c-x^2}{c-b}, \sqrt{b} \leq x \leq \sqrt{c} \\ 0, \quad otherwise \end{cases}$$

It can be seen that this tallies with the findings of Chou (2009). We could thus establish this result without actually using any mathematical rigour; we now proceed to generalize our idea in the next section.

4. The fmf of Functions of Fuzzy Numbers

Let $X = [a, b, c],\ (a,\ b,\ c > 0)$, be a fuzzy number. Let $F(X) = [F(a),\ F(b),\ F(c)]$, be the fuzzy number of any function $F(X)$. Suppose the fmf of X with membership as in (3.1). The left reference function $L(x)$ and the right reference function $R(x)$ are distribution function and a complementary distribution function respectively

(Baruah (2010b)). If the density functions of $L(x)$ and $1-R(x)$ are $f(x)$ and $g(x)$ respectively, i.e.

$$f(x)=\frac{d}{dx}\big(L(x)\big)=\frac{1}{b-a},\qquad a\le x\le b$$

and $g(x)=\frac{d}{dx}\big(1-R(x)\big)=\frac{1}{c-b},\ b\le x\le c.$

Let $y=F(x)$ or, $x=\phi(y)$ or, $\frac{d}{dy}(x)=\frac{d}{dy}\big(\phi(y)\big)=m(y)$, say. Replacing x by $\phi(y)$ in $f(x)$ and $g(x)$, we obtain $f(x)=\phi_1(y)$ and $g(x)=\phi_2(y)$, say. Then the membership function of $F(x)$ would be given by

$$\mu_{F(X)}(x)=\begin{cases}\int_{F(a)}^{x}\phi_1(y)m(y)dy, F(a)\le x\le F(b)\\ \int_{F(b)}^{x}\phi_2(y)m(y)dy, F(b)\le x\le F(c)\\ 0, \qquad otherwise\end{cases}$$

We now proceed to demonstrate this result with a few examples.

Example 1.

Let $X=[a,b,c],\ (a,b,c>0)$ be a tfn. Suppose the fmf of X with membership as in (3.1). Let $\sqrt[n]{X}=\left[\sqrt[n]{a},\sqrt[n]{b},\sqrt[n]{c}\right]$ be the fuzzy nth root of the tfn X. Let $y=\sqrt[n]{x}$ so that $x=y^n$, which implies $m(y)=ny^{n-1}$. Then the density functions $f(x)$ and $g(x)$ would be $f(x)=\frac{1}{b-a}=\phi_1(y)$ and $g(x)=\frac{1}{c-b}=\phi_2(y)$. Then the fmf of $\sqrt[n]{X}$ would be given by

$$\mu_{\sqrt[n]{X}}(x)=\begin{cases}\int_{\sqrt[n]{a}}^{x}\frac{1}{b-a}ny^{n-1}dy, \sqrt[n]{a}\le x\le\sqrt[n]{b}\\ \int_{\sqrt[n]{b}}^{x}\frac{1}{c-b}ny^{n-1}dy, \sqrt[n]{b}\le x\le\sqrt[n]{c}\\ 0, \qquad otherwise\end{cases}$$

$$=\begin{cases}\frac{x^n-a}{b-a}, & \sqrt[n]{a}\le x\le\sqrt[n]{b}\\ \frac{c-x^n}{c-b}, & \sqrt[n]{b}\le x\le\sqrt[n]{c}\\ 0, & otherwise\end{cases}$$

Example 2.

Let us consider a fuzzy number $X=[\sqrt{a},\ \sqrt{b},\ \sqrt{c}],(a,\ b,\ c>0)$, with fmf

$$\mu_X(x)=\begin{cases}\frac{x^2-a}{b-a}, \sqrt{a}\le x\le\sqrt{b}\\ \frac{c-x^2}{c-b}, \sqrt{b}\le x\le\sqrt{c}\\ 0, \qquad otherwise\end{cases}$$

Clearly, X here is not a tfn. Suppose, we are to find the fmf of this $\sqrt{X}$. Let $y=\sqrt{x}$ so that $x=y^2$, which implies $m(y)=\frac{d}{dy}(x)=2y$. The density functions would now be given by

$$f(x)=\frac{2x}{b-a}=\frac{2y^2}{b-a}=\phi_1(y)$$

and $g(x)=\frac{2x}{c-b}=\frac{2y^2}{c-b}=\phi_2(y).$

Then the fmf of $\sqrt{X}$ would be given by

$$\mu_{\sqrt{X}}(x)=\begin{cases}\frac{x^4-a}{b-a}, \sqrt[4]{a}\le x\le\sqrt[4]{b}\\ \frac{c-x^4}{c-b}, \sqrt[4]{b}\le x\le\sqrt[4]{c}\\ 0, \qquad otherwise\end{cases}$$

Indeed, this is nothing but fmf of $\sqrt[4]{X}$, when X is fuzzy.

Example 3.

Let $X=[a,b,c],\ (a,b,c>0)$ be a tfn. Suppose the fmf of X with membership as in (3.1). Let exp(X)=[exp(a), exp(b), exp(c)] be the fuzzy nth root of the tfn X. Let $y=\exp(x)$ so that $x=\ln(x)$, which implies $m(y)=\frac{1}{y}$. Then the density functions f(x) and g(x) would be $f(x)=\frac{1}{b-a}=\phi_1(y)$ and $g(x)=\frac{1}{c-b}=\phi_2(y)$. Then the fmf of exp(X) would be given by

$$\mu_{\exp(X)}=\begin{cases}\int_{\exp(a)}^{x}\frac{1}{b-a}\left(\frac{1}{y}\right)dy, \exp(a)\le x\le\exp(b)\\ \int_{\exp(b)}^{x}\frac{1}{c-b}\left(\frac{1}{y}\right)dy, \exp(b)\le x\le\exp(c)\\ 0, \qquad otherwise\end{cases}$$

$$= \begin{cases} \dfrac{ln(x) - a}{b - a}, \exp(a) \leq x \leq \exp(b) \\ \dfrac{c - \ln(x)}{c - b}, \exp(b) \leq x \leq \exp(c) \\ 0, \quad otherwise \end{cases}$$

Example 4.

Let $X = [a, b, c],\ (a, b, c > 0)$ be a tfn. Let, $X^{-1} = [c^{-1}, b^{-1}, a^{-1}]$ be the fuzzy inverse of the tfn X. Then the fmf of X^{-1} would similarly be found as,

$$\mu_{X^{-1}}(x) = \begin{cases} \dfrac{1 - cx}{(b - c)x}, \dfrac{1}{c} \leq x \leq \dfrac{1}{b} \\ \dfrac{1 - ax}{(b - a)x}, \dfrac{1}{b} \leq x \leq \dfrac{1}{a} \\ 0 \quad , \ otherwise \end{cases}$$

5. Conclusion

The standard method of α-cuts to the membership of a fuzzy number does not always yield result. We have demonstrated that an assumption that the Dubois-Prade left reference function is a distribution function and that the right reference function is a complementary distribution function leads to the finding membership in a very simpler way.

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References

[1]. Baruah, Hemanta K.; (1999), Set Superimposition and Its Applications to the Theory of Fuzzy Sets, *Journal of the Assam Science Society*, Vol. 40, Nos. 1 & 2, 25-31.

[2]. Baruah Hemanta K.; (2010a); Construction of The Membership Function of a Fuzzy Number, ICIC Express Letters (Accepted for Publication: to appear in February 2011).

[3] Baruah Hemanta K.; (2010b); The Mathematics of Fuzziness: Myths and Realities, Lambert Academic Publishing, Saarbruken, Germeny.

[4]. Chou, Chien-Chang; (2009), The Square Roots of Triangular Fuzzy Number, ICIC Express Letters. Vol. 3, Nos. 2, pp. 207-212.

[5]. de Barra, G.; (1987), Measure Theory and Integration, Wiley Eastern Limited, New Delhi.

[6]. Kaufmann A., and M. M. Gupta; (1984), Introduction to Fuzzy Arithmetic, Theory and Applications, Van Nostrand Reinhold Co. Inc., Wokingham, Berkshire.

[7]. Loeve M., (1977), *Probability Theory*, Vol.I, Springer Verlag, New York.

FUZZY ARITHMETIC WITHOUT USING THE METHOD OF α - CUTS

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Abstract: In this article, an alternative method to evaluate the arithmetic operations on fuzzy number has been developed, on the assumption that the Dubois-Prade left and right reference functions of a fuzzy number are *distribution function* and *complementary distribution function* respectively. Using the method, the arithmetic operations of fuzzy numbers can be done in a very simple way. This alternative method has been demonstrated with the help of numerical examples.

Key words and phrases: Fuzzy membership function, Dubois-Prade reference function, distribution function, Set superimposition, Glivenko-Cantelli theorem.

1. INTRODUCTION

The standard method of α-cuts to the membership of fuzzy number does not always yield results. For example, the method of α-cuts fails to find the fuzzy membership function (fmf) of even the simple function $\sqrt{X}$ when X is fuzzy. Indeed, for $\sqrt{X}$ in particular, Chou (2009) has forwarded a method of finding the fmf for a triangular fuzzy number X. We shall in this article, put forward an alternative method for dealing with the arithmetic of fuzzy numbers which are not necessarily triangular.

Dubois and Prade (see e.g. Kaufmann and Gupta (1984)) have defined a fuzzy number $X=[a,b,c]$ with membership function

$$\mu_X(x)=\begin{cases} L(x), & a\le x\le b \\ R(x), & b\le x\le c \\ 0, & otherwise \end{cases} \qquad (1.1)$$

$L(x)$ being continuous non-decreasing function in the interval [a, b], and $R(x)$ being a continuous non-increasing function in the interval [b, c], with $L(a)=R(c)=0$ and $L(b)=R(b)=1$. Dubois and Prade named $L(x)$ as left reference function and $R(x)$ as right reference function of the concerned fuzzy number. A continuous non-decreasing function of this type is also called a distribution function with reference to a Lebesgue-Stieltjes measure (de Barra (1987). pp-156).

In this article, we are going to demonstrate the easiness of applying our method in evaluating the arithmetic of fuzzy numbers if start from the simple assumption that the Dubois-Prade left reference function is a *distribution function*, and similarly the Dubois-Prade right reference function is a *complementary distribution function*. Accordingly, the functions $L(x)$ and $(1-R(x))$ would have to be associated with densities $\frac{d}{dx}(L(x))$ and $\frac{d}{dx}(1-R(x))$ in [a,b] and [b,c] respectively (Baruah (2010 a, b)).

2. SUPERIMPOSITION OF SETS

The superimposition of sets defined by Baruah (1999), and later used successfully in recognizing periodic patterns (Mahanta et al. (2008)), the operation of set superimposition is defined as follows: if the set A is superimposed over the set B, we get

$$A(S)\,B = (A-B) \cup (A \cap B)^{(2)} \cup (B-A) \qquad (2.1)$$

where S represents the operation of superimposition, and $(A \cap B)^{(2)}$ represents the elements of $(A \cap B)$ occurring twice. It can be seen that for two intervals $[a_1, b_1]$ and $[a_2, b_2]$ superimposed gives

$$[a_1, b_1]\,(S)\,[a_2, b_2]$$

$$= [a_{(1)}, a_{(2)}] \cup [a_{(2)}, b_{(1)}]^{(2)} \cup [b_{(1)}, b_{(2)}]$$

where $a_{(1)} = \min(a_1, a_2)$, $a_{(2)} = \max(a_1, a_2)$, $b_{(1)} = \min(b_1, b_2)$, and $b_{(2)} = \max(b_1, b_2)$.

Indeed, in the same way if $[a_1, b_1]^{(1/2)}$ and $[a_2, b_2]^{(1/2)}$ represent two uniformly fuzzy intervals both with membership value equal to half everywhere, superimposition of $[a_1, b_1]^{(1/2)}$ and $[a_2, b_2]^{(1/2)}$ would give rise to

$$[a_1, b_1]^{(1/2)}\,(S)\,[a_2, b_2]^{(1/2)}$$

$$= [a_{(1)}, a_{(2)}]^{(1/2)} \cup [a_{(2)}, b_{(1)}]^{(1)} \cup [b_{(1)}, b_{(2)}]^{(1/2)}. \qquad (2.2)$$

So for n fuzzy intervals $[a_1, b_1]^{(1/n)}$, $[a_2, b_2]^{(1/n)} \ldots [a_n, b_n]^{(1/n)}$ all with membership value equal to 1/n everywhere,

$$[a_1, b_1]^{(1/n)}(S)\,[a_2, b_2]^{(1/n)}(S) \ldots\ldots (S)\,[a_n, b_n]^{(1/n)}$$

$$= [a_{(1)}, a_{(2)}]^{(1/n)} \cup [a_{(2)}, a_{(3)}]^{(2/n)} \cup \ldots\ldots\ldots \cup [a_{(n-1)}, a_{(n)}]^{((n-1)/n)} \cup [a_{(n)}, b_{(1)}]^{(1)} \cup [b_{(1)}, b_{(2)}]^{((n-1)/n)} \cup \ldots\ldots\ldots \cup [b_{(n-2)}, b_{(n-1)}]^{(2/n)} \cup [b_{(n-1)}, b_{(n)}]^{(1/n)}, \qquad (2.3)$$

where, for example, $[b_{(1)}, b_{(2)}]^{((n-1)/n)}$ represents the uniformly fuzzy interval $[b_{(1)}, b_{(2)}]$ with membership ((n-1) /n) in the entire interval, $a_{(1)}, a_{(2)}, \ldots\ldots\ldots, a_{(n)}$ being values of $a_1, a_2, \ldots\ldots\ldots, a_n$ arranged in increasing order of magnitude, and $b_{(1)}, b_{(2)}, \ldots\ldots\ldots, b_{(n)}$ being values of $b_1, b_2, \ldots\ldots\ldots, b_n$ arranged in increasing order of magnitude.

We now define a *random* vector $X = (X_1, X_2, \ldots\ldots, X_n)$ as a family of X_k, k = 1, 2,……, n, with every $X_k$inducing a sub-σ field so that X is measurable. Let $(x_1, x_2 \ldots x_n)$ be a particular realization of X, and let $X_{(k)}$ realize the value $x_{(k)}$ where $x_{(1)}, x_{(2)}, \ldots, x_{(n)}$ are *ordered* values of $x_1, x_2, \ldots\ldots, x_n$in increasing order of magnitude. Further let the sub-σ fields induced by $X_k$be independent and identical. Define now

$$\Phi_n(x) = 0, \text{ if } x < x_{(1)},$$

$$= (r-1)/n, \text{ if } x_{(r-1)} \leq x \leq x_{(r)}, r = 2, 3, \ldots, n,$$

$$= 1, \text{ if } x \geq x_{(n)}; \qquad (2.4)$$

$\Phi_n(x)$ here is an empirical distribution function of a theoretical distribution function $\Phi(x)$.

As there is a one to one correspondence between a Lebesgue-Stieltjes measure and the distribution function, we would have

$$\Pi(a, b) = \Phi(b) - \Phi(a) \qquad (2.5)$$

whereΠ is a measure in (Ω, A, Π), *A* being the σ- field common to every x_k.

Now the Glivenko-Cantelli theorem (see e.g. Loeve (1977), pp-20) states that

$\Phi_n(x)$ converges to $\Phi(x)$ uniformly in x. This means,

$$\sup \left| \Phi_n(x) - \Phi(x) \right| \to 0 \qquad (2.6)$$

Observe that (r-1)/n in (2.4), for $x_{(r-1)} \leq x \leq x_{(r)}$, are membership values of $[a_{(r-1)}, a_{(r)}]^{((r-1)/n)}$ and $[b_{(n-r+1)}, b_{(n-r)}]^{((r-1)/n)}$ in (2.3), for r = 2, 3, …, n. Indeed this fact found from superimposition of uniformly fuzzy sets has led us to look into the possibility that there could possibly be a link between distribution functions and fuzzy membership.

In the sections 3, 4, 5 and 6 we are going to discuss the arithmetic of fuzzy numbers.

3. ADDITION OF FUZZY NUMBERS

Consider $X = [a,b,c]$ and $Y = [p,q,r]$ be two triangular fuzzy numbers. Suppose $Z = X + Y = [a+p, b+q, c+r]$ be the fuzzy number of $X + Y$. Let the fmf of X and Y be $\mu_X(x)$ and $\mu_Y(y)$ as mentioned below

$$\mu_X(x) = \begin{cases} L(x), & a \leq x \leq b \\ R(x), & b \leq x \leq c \\ 0, & otherwise \end{cases} \qquad (3.1)$$

and

$$\mu_Y(y) = \begin{cases} L(y), & a \leq y \leq b \\ R(y), & b \leq y \leq c \\ 0, & otherwise \end{cases} \qquad (3.2)$$

where $L(x)$ and $L(y)$ are the left reference functions and $R(x)$ and $R(y)$ are the right reference functions respectively. We assume that $L(x)$ and $L(y)$ are *distribution function* and $R(x)$ and $R(y)$ are *complementary distribution function*. Accordingly, there would exist some density functions for the distribution functions $L(x)$ and $(1-R(x))$. Say,

$$f(x) = \frac{d}{dx}(L(x)), a \leq x \leq b$$

and

$$g(x) = \frac{d}{dx}(1-R(x)), b \leq x \leq c$$

We start with equating $L(x)$ with $L(y)$, and $R(x)$ with $R(y)$. And so, we obtain $y = \phi_1(x)$ and $y = \phi_2(x)$ respectively. Let $z = x + y$, so we have $z = x + \phi_1(x)$ and $z = x + \phi_2(x)$, so that $x = \psi_1(z)$ and $x = \psi_2(z)$, say. Replacing x by $\psi_1(z)$ in $f(x)$, and by $\psi_2(z)$ in $g(x)$, we obtain $f(x) = \eta_1(z)$ and $g(x) = \eta_2(z)$ say.

Now let, $$\frac{dx}{dz} = \frac{d}{dz}(\psi_1(z)) = m_1(z)$$

and $$\frac{dx}{dz} = \frac{d}{dz}(\psi_2(z)) = m_2(z)$$

The distribution function for $X + Y$, would now be given by

$$\int_{a+p}^{x} \eta_1(z) m_1(z)\, dz,\ a+p \leq x \leq b+q$$

and the complementary distribution function would be given by

$$1 - \int_{b+q}^{x} \eta_2(z) m_2(z)\, dz,\ b+q \leq x \leq c+r$$

We claim that this distribution function and the complementary distribution function constitute the fuzzy membership function of $X + Y$ as,

$$\mu_{X+Y}(x) = \begin{cases} \int_{a+p}^{x} \eta_1(z) m_1(z)\, dz, & a+p \leq x \leq b+q \\ 1 - \int_{b+q}^{x} \eta_2(z) m_2(z)\, dz, & b+q \leq x \leq c+r \\ 0 & ,otherwise \end{cases}$$

4. SUBTRACTION OF FUZZY NUMBERS

Let $X=[a,b,c]$ and $Y=[p,q,r]$ be two fuzzy numbers with fuzzy membership function as in (3.1) and (3.2). Suppose $Z=X-Y$. Then the fuzzy membership function of $Z=X-Y$ would be given by $Z=X+(-Y)$.

Suppose $(-Y)=[-r,-q,-p]$ be the fuzzy number of $(-Y)$. We assume that the Dubois-Prade reference functions $L(y)$ and $R(y)$ as distribution and complementary distribution function respectively. Accordingly, there would exist some density functions for the distribution functions $L(y)$ and $(1-R(y))$. Say,

$$f(y)=\frac{d}{dy}(L(y)),\ p\le y\le q$$

and $$g(y)=\frac{d}{dy}(1-R(y)),\ q\le y\le r$$

Let $t=-y$ so that $\frac{dy}{dt}=-1=m(t)$, say. Replacing $y=-t$ in $f(y)$ and $g(y)$, we obtain $f(y)=\eta_1(t)$ and $g(y)=\eta_2(t)$, say. Then the fmf of $(-Y)$ would be given by

$$\mu_{-Y}(y)=\begin{cases}\int_{-r}^{y}\eta_2(t)m(t)dt, & -r\le y\le -q\\ 1-\int_{-q}^{y}\eta_1(t)m(t)dt, & -q\le y\le -p\\ 0 & ,otherwise\end{cases}$$

Then we can easily find the fmf of $X-Y$ by addition of fuzzy numbers X and $(-Y)$ as described in the earlier section.

5. MULTIPLICATION OF FUZZY NUMBERS

Let $X=[a,b,c]$, $(a,b,c>0)$ and $Y=[p,q,r]$, $(p,q,r>0)$ be two triangular fuzzy numbers with fuzzy membership function as in (3.1) and (3.2). Suppose $Z=X.Y=[a.p,b.q,c.r]$ be the fuzzy number of $X.Y$. $L(x)$ and $L(y)$ are the left reference functions and $R(x)$ and $R(y)$ are the right reference functions respectively. We assume that $L(x)$ and $L(y)$ are distribution function and $R(x)$ and $R(y)$ are complementary distribution function. Accordingly, there would exist some density functions for the distribution functions $L(x)$ and $(1-R(x))$. Say,

$$f(x)=\frac{d}{dx}(L(x)),\ a\le x\le b$$

and $$g(x)=\frac{d}{dx}(1-R(x)),\ b\le x\le c$$

We again start with equating $L(x)$ with $L(y)$, and $R(x)$ with $R(y)$. And so, we obtain $y=\phi_1(x)$ and $y=\phi_2(x)$ respectively. Let $z=x.y$, so we have $z=x.\phi_1(x)$ and $z=x.\phi_2(x)$, so that $x=\psi_1(z)$ and $x=\psi_2(z)$, say. Replacing x by $\psi_1(z)$ in $f(x)$, and by $\psi_2(z)$ in $g(x)$, we obtain $f(x)=\eta_1(z)$ and $g(x)=\eta_2(z)$ say.

Now let, $$\frac{dx}{dz}=\frac{d}{dz}(\psi_1(z))=m_1(z)$$

and $$\frac{dx}{dz}=\frac{d}{dz}(\psi_2(z))=m_2(z)$$

The distribution function for $X.Y$, would now be given by

$$\int_{ap}^{x} \eta_1(z) m_1(z) dz,\ ap \le x \le bq$$

and the complementary distribution function would be given by

$$1 - \int_{bq}^{x} \eta_2(z) m_2(z) dz,\ bq \le x \le cr$$

We are claiming that this distribution function and the complementary distribution function constitute the fuzzy membership function of $X.Y$ as,

$$\mu_{XY}(x) = \begin{cases} \int_{ap}^{x} \eta_1(z) m_1(z) dz, & ap \le x \le bq \\ 1 - \int_{bq}^{x} \eta_2(z) m_2(z) dz, & bq \le x \le cr \\ 0 & ,otherwise \end{cases}$$

6. DIVISION OF FUZZY NUMBERS

Let $X = [a,b,c]$, $(a,b,c > 0)$ and $Y = [p,q,r]$, $(p,q,r > 0)$ be two triangular fuzzy numbers with fuzzy membership function as in (3.1) and (3.2). Suppose $Z = \frac{X}{Y}$. Then the fuzzy membership function of $Z = \frac{X}{Y}$ would be given by $Z = X.Y^{-1}$.

At first, we have to find the fmf of Y^{-1}. Suppose $Y^{-1} = [r^{-1}, q^{-1}, p^{-1}]$ be the fuzzy number of Y^{-1}. We assume that the Dubois-Prade reference functions $L(y)$ and $R(y)$ as distribution and complementary distribution function respectively. Accordingly, there would exist some density functions for the distribution functions $L(y)$ and $(1 - R(y))$. Say,

$$f(y) = \frac{d}{dy}(L(y)),\ p \le y \le q$$

and $g(y) = \frac{d}{dy}(1 - R(y)),\ q \le y \le r$

Let $t = y^{-1}$ so that $\frac{dy}{dt} = -\frac{1}{t^2} = m(t)$, say. Replacing $y = t^{-1}$ in $f(y)$ and $g(y)$, we obtain $f(y) = \eta_1(t)$ and $g(y) = \eta_2(t)$, say. Then the fmf of (Y^{-1}) would be given by

$$\mu_{Y^{-1}}(y) = \begin{cases} \int_{r^{-1}}^{y} \eta_2(t) m(t) dt, & r^{-1} \le y \le q^{-1} \\ 1 - \int_{q^{-1}}^{y} \eta_1(t) m(t) dt, & q^{-1} \le y \le p^{-1} \\ 0 & ,otherwise \end{cases}$$

Next, we can easily find the fmf of $\frac{X}{Y}$ by multiplication of fuzzy numbers X and Y^{-1} as described in the earlier section.

In the next section we are going to cite some numerical examples for the above discussed methods.

7. NUMERICAL EXAMPLES

Example 1:

Let $X = [1,2,4]$ and $Y = [3,5,6]$ be two triangular fuzzy numbers with fmf

$$\mu_X(x) = \begin{cases} x - 1, & 1 \le x \le 2 \\ \frac{4 - x}{2}, & 2 \le x \le 4 \\ 0 & ,otherwise \end{cases} \qquad (7.1)$$

And $$\mu_Y(y)=\begin{cases}\frac{y-3}{2}, & 3\le y\le 5\\ 6-y, & 5\le y\le 6\\ 0 & ,otherwise\end{cases} \qquad (7.2)$$

Here $X+Y=[4,7,10]$. Equating the distribution function and complementary distribution function, we obtain $y=\phi_1(x)=2x+1$ and $y=\phi_2(x)=\frac{x+8}{2}$. Let $z=x+y$, so we shall have

$z=x+\phi_1(x)=3x+1$ and $z=x+\phi_2(x)=\frac{3x+8}{2}$,

so that $x=\psi_1(z)=\frac{z-1}{3}$ and $x=\psi_2(z)=\frac{2z-8}{3}$, respectively. Replacing x by $\psi_1(z)$ and $\psi_2(z)$ in the density functions $f(x)$ and $g(x)$ respectively, we have $f(x)=1=\eta_1(z)$ and $g(x)=\frac{1}{2}=\eta_2(z)$.

Now $m_1(z)=\frac{d}{dz}(\psi_1(z))=\frac{1}{3}$

and $m_2(z)=\frac{d}{dz}(\psi_2(z))=\frac{2}{3}$.

Then the fmf of $X+Y$ would be given by,

$$\mu_{X+Y}(x)=\begin{cases}\frac{x-4}{3}, & 4\le x\le 7\\ \frac{10-x}{3}, & 7\le x\le 10\\ 0 & ,otherwise\end{cases}$$

Example 2:

Let $X=[1,2,4]$ and $Y=[3,5,6]$ be two triangular fuzzy numbers with membership functions as in (7.1) and (7.2). Suppose, $Z=X-Y$ or $Z=X+(-Y)$.

Now, $-Y=[-6,-5,-3]$ be the fuzzy number of $(-Y)$. Let $t=-y$ so that $y=-t$, which implies $m(t)=-1$. Then the density function $f(y)$ and $g(y)$ would be, say,

$f(y)=\frac{d}{dy}\left(\frac{y-3}{2}\right)=\frac{1}{2}=\eta_1(t),\ 3\le y\le 5$ and

$g(y)=\frac{d}{dy}(1-(6-y))=1=\eta_2(t),\ 5\le y\le 6$.

Then the fmf of $(-Y)$ would be given by

$$\mu_{-Y}(y)=\begin{cases}\frac{6+y}{6-5}, & -6\le y\le -5\\ \frac{3+y}{3-5}, & -5\le y\le -3\\ 0 & ,otherwise\end{cases}$$

Then by addition of fuzzy numbers $X=[1,2,4]$ and $(-Y)=[-6,-5,-3]$ the fmf of $X-Y$ is given by,

$$\mu_{X+(-Y)}(x)=\begin{cases}\frac{x+5}{2}, & -5\le x\le -3\\ \frac{1-x}{4}, & -3\le x\le 1\\ 0 & ,otherwise\end{cases}$$

Example 3:

Let $X=[1,2,4]$ and $Y=[3,5,6]$ be two triangular fuzzy numbers with membership functions as in (7.1) and (7.2). Suppose, $X.Y=[3,10,24]$ be the fuzzy number of $X.Y$. Equating the distribution function and complementary distribution function, we obtain $y=\phi_1(x)=2x+1$ and $y=\phi_2(x)=\frac{x+8}{2}$.

Let $z = x.y$, so we shall have $z = x.\phi_1(x) = 2x^2 + x$ and

$$z = x.\phi_2(x) = \frac{8x + x^2}{2}$$, so that

$$x = \psi_1(z) = \frac{-1 \pm \sqrt{1+8z}}{4}$$

and $x = \psi_2(z) = -4 \pm \sqrt{16+2z}$. Replacing x by $\psi_1(z)$ and $\psi_2(z)$ in the density functions $f(x)$ and $g(x)$ respectively, we have $f(x) = 1 = \eta_1(z)$ and $g(x) = \frac{1}{2} = \eta_2(z)$.

Now $m_1(z) = \frac{d}{dz}(\psi_1(z))$ and $m_2(z) = \frac{d}{dz}(\psi_2(z))$.

Then the fmf of $X.Y$ would be given by

$$\mu_{X.Y}(x) = \begin{cases} \frac{\sqrt{1+8x}-5}{4}, & 3 \le x \le 10 \\ \frac{8-\sqrt{16+2x}}{2}, & 10 \le x \le 24 \\ 0 \quad ,otherwise \end{cases}$$

Example 4:

Let $X = [1,2,4]$ and $Y = [3,5,6]$ be two triangular fuzzy numbers with membership functions as in (7.1) and (7.2). Suppose, $Z = \frac{X}{Y}$ or $Z = X.Y^{-1}$. Then the fmf of $Y^{-1} = [6^{-1}, 5^{-1}, 3^{-1}]$ is given as

$$\mu_{\frac{1}{Y}}(y) = \begin{cases} \frac{6-\frac{1}{y}}{6-5}, & \frac{1}{6} \le y \le \frac{1}{5} \\ \frac{\frac{1}{y}-3}{5-3}, & \frac{1}{5} \le y \le \frac{1}{3} \\ 0 \quad ,otherwise \end{cases}$$

Then by multiplication of fuzzy numbers $X = [1,2,4]$ and $Y^{-1} = [6^{-1}, 5^{-1}, 3^{-1}]$ the fuzzy membership function of $\frac{X}{Y}$ would be given by,

$$\mu_{\frac{X}{Y}}(x) = \begin{cases} \frac{6x-1}{x+1}, & \frac{1}{6} \le x \le \frac{2}{5} \\ \frac{4-3x}{2(x+1)}, & \frac{2}{5} \le x \le \frac{4}{3} \\ 0 \quad ,otherwise \end{cases}$$

Example 5:

Let $X = [2,4,5]$ be a triangular fuzzy number and $Y = [4,16,25]$ which is a non-triangular fuzzy number with membership functions respectively as,

$$\mu_X(x) = \begin{cases} \frac{x-2}{2}, & 2 \le x \le 4 \\ 5-x, & 4 \le x \le 5 \\ 0 \quad ,otherwise \end{cases}$$

and $$\mu_Y(y) = \begin{cases} \frac{\sqrt{y}-2}{2}, & 4 \le y \le 16 \\ 5-\sqrt{y}, & 16 \le y \le 25 \\ 0 \quad ,otherwise \end{cases}$$

We can find the fmf of $X+Y$ which is given by,

$$\mu_{X+Y}(x) = \begin{cases} \dfrac{\sqrt{1+4x}-2}{4}, & 6 \le x \le 20 \\ \dfrac{11-\sqrt{1+4x}}{2}, & 20 \le x \le 30 \\ 0 & ,otherwise \end{cases}$$

All four demonstrations above can be verified to be true, using the method of α -cuts.

9. CONCLUSION

The standard method of α -cuts to the membership of a fuzzy number does not always yield results. We have demonstrated that an assumption that the Dubois-Prade left reference function is a distribution function and that the right reference function is a complementary distribution function leads to a very simple way of dealing with fuzzy arithmetic. Further, this alternative method can be utilized in the cases where the method of α -cuts fails, e.g. in finding the fmf of $\sqrt{X}$.

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REFERENCES

[1]. Baruah, Hemanta K.; (1999), "Set Superimposition and Its Applications to the Theory of Fuzzy Sets", Journal of the Assam Science Society, Vol. 40, Nos. 1 & 2, 25-31.

[2]. Baruah, Hemanta K.; (2010 a), "Construction of The Membership Function of a Fuzzy Number", ICIC Express Letters (Accepted for Publication: to appear in February, 2011).

[3]. Baruah, Hemanta K.; (2010 b), "The Mathematics of Fuzziness: Myths and Realities", Lambert Academic Publishing, Saarbrucken, Germany.

[4]. Chou, Chien-Chang; (2009), "The Square Roots of Triangular Fuzzy Number", ICIC Express Letters. Vol. 3, Nos. 2, pp. 207-212.

[5]. de Barra, G.; (1987), "Measure Theory and Integration", Wiley Eastern Limited, New Delhi.

[6]. Kaufmann A., and M. M. Gupta; (1984), "Introduction to Fuzzy Arithmetic, Theory and Applications", Van Nostrand Reinhold Co. Inc., Wokingham, Berkshire.

[7]. Loeve M., (1977), "Probability Theory", Vol.I, Springer Verlag, New York.

[8]. Mahanta, Anjana K., Fokrul A. Mazarbhuiya and Hemanta K. Baruah; (2008), "Finding Calendar Based Periodic Patterns", Pattern Recognition Letters, 29 (9), 1274-1284.

NHPP and S-Shaped Models for Testing the Software Failure Process

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***Abstract*:** Non-homogeneous Poisson process plays an important role in software and hardware reliability engineering. In many realistic situations there are two or more change points in **NHPP** models. In the software reliability, the nature of the failure data is affected by many factors, such as testing environment, testing strategy, and resource allocation. These factors are stable through the entire process of reliability analysis. In this paper, we test the different change points according to their existence by using some test statistics.

***Keywords*:** Change points, reliability and S -Shaped Model.

1. Introduction

NHPP models play an important role in software and hardware reliability. Musa et.al (1987), Xie (1991), Pham (1999) and Singpurwalla and Wilson (1999) among others developed the different software reliability models. The first **NHPP** software reliability model is proposed by Goel and Okumoto (1979), they assumed that the software failure intensity is proportional to the expected number of undetected failures. Musa anf Okumoto (1984) give the log-arithmetic Poisson execution time model.

Zhao1993), first considered the change-point problem in software reliability. He modified the Jelinski-Moranda model (1972) to estimate the location of change point. Chang (2001) and Zou(2003) uses some useful **NHPP** software reliability models with change point. Shyur(2003) incorporated both imperfect debugging and change-point problem into **NHPP** model. In the **NHPP** model, there is only one change-point and the unknown change point can be estimated by the maximum likelihood method or the LS method. However, in many realistic situations, the change-point is unknown. Chang (2001) shows that the two change-points oppose each other. Chen and Gupta (2001) considered a problem of multiple change points. In the present paper, we considered change-point detections. At first, we give the **NHPP** models with multiple change points and maximum likelihood method is used to estimate the change points and other parameters of model. To test the existence of change-point(s) the test statistics is proposed.

2. NHPP Models with Change Points

Many NHPP models are very useful to describe the software failure process. In the present paper the delayed S – Shaped model with one change point is considered, and after that delayed S-Shaped model with multiple change points are considered.

2.1 S-Shaped Model

Software failure processes are called as fault counting process. Let $\{N(t);t>0\}$ is considered as the cumulative number of software failure time by t. The N (t) is called as **NHPP** with mean value function m (t) and failure intensity $\lambda(t)$. Geol and Okumoto (1979) assume that the software failure intensity $\lambda_0(t)$ is proportional to the expected number of undetected failure i.e.,

$$\lambda_0(t)=\frac{d\,m_0(t)}{dt}=b[a-m_0(t)] \quad \ldots \quad (1)$$

Where, ***a*** is initial number of faults contained in the software and ***b*** is called as the fault detection rate, the mean value function and intensity function are,

$$m_0(t)=a\left(1-e^{-bt}\right) \quad \ldots \quad (2)$$

And

$$\lambda_0(t)=ab^2te^{-\beta t} \;;\; \alpha>0;\beta\in R \quad \ldots \quad (3)$$

Suppose that ***n*** software failures are obtained and software process lasted at time ***T***. Let $0<t_1<t_2<\ldots\ldots\ldots\leq t_n<T$ in which failures are

observed. The log-likelihood functions of the observed data are,

$$\begin{aligned}\log L_0(a,b) &= \sum_{i=1}^{n} \log \lambda_0(t_i) - m_0(T) \\ &= \sum_{i=1}^{n}\left[ab^2 t_i e^{-bt_i} - m_0(T)\right] \\ &= n(\log a + 2\log b + T - bT) - a\left(1 - e^{-bT}\right)\end{aligned}$$

The maximum likelihood estimator of parameters *a*, *b* are obtained by solving the following two equations,

$$\frac{n}{a} = \left(1 - e^{-bT}\right) \quad \ldots \quad (4)$$

$$\frac{2}{b} - \sum_{i=1}^{n}(T - t_i) - \frac{nT}{\left(1 - e^{-bT}\right)} = 0 \quad \ldots \quad (5)$$

3. S- Shaped Model with Change-Points

Whenever the software testing process is going on, the nature of the failure data can be affected many factors such as testing environment, testing strategy, resources allocation and so on. In this case, it is good to use a change-point method in reliability analysis.

The fault detection rate b is not a constant; it is assumed to have a change-point. Therefore, the fault detection rate at testing time t can be defined as,

$$b(t) = \begin{cases} b_1 & ; 0 \le t \le \eta \\ b_2 & ; t > \eta \end{cases} \quad \ldots \quad (6)$$

The η is the change points b_1 and b_2 are the fault detection rates before and after the change points. If $b_1 = b_2$ the change point model is equivalent to delayed S-Shaped model. Under the assumption,

$$\lambda_1(t) = \frac{dm_1(t)}{dt} = b(t)\left[a - m_1(t)\right]$$

The mean value function and intensity function can be expressed as,

$$m_1(t) = \begin{cases} a\left(1 - e^{-bt}\right) & ; 0 \le t \le \eta \\ a\left[1 - e^{-b_1\eta - b_2(t-\eta)}\right] & ; t > \eta \end{cases}$$

$$\ldots \quad (7)$$

And

$$\lambda_1(t) = \begin{cases} ab_1^2 t\, e^{-b_1 t} & ; 0 \le t \le \eta \\ a\, b_2^2 t\, e^{-b_1\eta - b_2(t-\eta)} & ; t > \eta \end{cases}$$

$$\ldots \quad (8)$$

The log-likelihood function is,

$$\log L_1(\eta, a, b_1, b_2) = \log a + ae^{-b_1\eta - b_2(t-\eta)} + \sum_{i=1}^{N(\eta)}\left[\log(ab_1^2) + t_i - b_1 t_i\right] + \sum_{i=N(\eta)+1}^{n}\left[\log(ab_2^2) + t_i - b_1\eta - b_2(t_i - \eta)\right]$$

By Ngugen et.al (1984), assumes the log-likelihood function tends to infinity as the change point η tends to failure time t_n from below. Hence, the estimate value of η cannot be obtained by maximizing the log-likelihood function over $[0, T]$. Wang and Wang (2005) restrict the change point in the interval $[t_2, t_{n-1}]$. Here, we considered the change point lying in the interval $[t_2, t_{n-1}]$. Therefore the estimates value of the parameters $\hat{\eta}, \hat{a}, \hat{b}_1, \hat{b}_2$ are,

$$\log L_1(\hat{\eta}, \hat{a}, \hat{b}_1, \hat{b}_2) = \max_{\eta \in [t_2, t_n], a, b_1, b_2} \log L_1(\eta, a, b_1, b_2)$$

$$= \max_{\eta \in [t_2 \ldots\ldots t_{n-1}, t_3 \ldots\ldots t_{n-1}], a, b_1, b_2} \log L_1(\eta, a, b_1, b_2)$$

If $\eta = t_m, 2 \le m \le n-1$, then the estimates of parameter of a, b_1, b_2 are obtained by following equations,

(1) For parameter **a**,

$$\frac{n}{a} = 1 - e^{-b_1 t_m - b_2(T - t_m)} \quad \ldots \quad (9)$$

(2) For parameter b_1,

$$\frac{2m}{b_1} + \sum_{i=1}^{m}(t_m - t_i) - \frac{nt_m}{\left(1 - e^{-b_1 t_m - b_2(T - t_m)}\right)} = 0 \quad \ldots \quad (10)$$

(3) For parameter b_2,

$$\frac{2(n-m)}{b_2} + 2m(T - t_m) + \sum_{i=1}^{m}(t_m - t_i) - \frac{n(T - t_m)}{\left(1 - e^{-b_1 t_m - b_2(T - t_m)}\right)} = 0$$

$$\ldots. \quad (11)$$

Similarly, optimal solution of a_m, b_{1m}, b_{2m} are obtained as,

(1) For parameter a_m,

$$\frac{n}{a_m} = 1 - e^{-b_1 t_m - b_2 (T - t_m)} \quad \ldots \quad (12)$$

(2) For parameter b_{1m},

$$\frac{2m}{b_{1m}} + \sum_{i=1}^{m} (t_m - t_i) - \frac{n t_m}{\left(1 - e^{-b_{1m} t_m - b_{2m}(T - t_m)}\right)} = 0 \quad \ldots \quad (13)$$

(3) For parameter b_{2m}

$$\frac{2(n-m)}{b_{2m}} + 2m(T - t_m) + \sum_{i=1}^{m} (t_m - t_i) - \frac{n(T - t_m)}{\left(1 - e^{-b_{1m} t_m - b_{2m}(T - t_m)}\right)} = 0 \quad \ldots \quad (14)$$

In addition, if $\eta = t_m^-; 3 \le m \le n-1$, the estimate of a, b_1 and b_2 can be obtained accordingly,

$$\frac{n}{a} = 1 - e^{-b_1 t_m - b_2 (T - t_m)} \quad \ldots \quad (15)$$

$$\frac{2m-1}{b_1} + \sum_{i=1}^{m} (t_m - t_i) - \frac{n t_m}{\left(1 - e^{-b_{1m} t_m - b_{2m}(T - t_m)}\right)} = 0 \quad \ldots \quad (16)$$

$$\frac{2(n-m)+1}{b_2} + 2m(T - t_m) + \sum_{i=1}^{m} (t_m - t_i) - \frac{n(T - t_m)}{\left(1 - e^{-b_{1m} t_m - b_{2m}(T - t_m)}\right)} = 0 \quad \ldots \quad (17)$$

The optimal solution is denoted by $a_m^-, b_{1m}^-, b_{2m}^-$ respectively. Then the estimates $\hat{\eta}, \hat{a}, \hat{b}_1, \hat{b}_2$ can be obtained by comparing the values of $\log L_1(t_m, a_{m,}, b_{1m}, b_{2m}); m = 2, \ldots, n-1$ and $\log L_1(t_m, a_{m,}^-, b_{1m}^-, b_{2m}^-); m = 3, \ldots, n-1$.

Here, we present S-Shaped model with k change point, fault detection rate is given by,

$$b(t) = \{b_1, b_2, \ldots b_{k+1}; \ 0 \le t \le \eta_1; \ldots; t > \eta_k$$

Where, $\eta_1, \eta_2, \ldots, \eta_k$ is change point and $b_1, b_2, \ldots, b_{k+1}$ are fault detection rates. The mean value function and intensity function are as follows:

$$m_k(t) = \begin{cases} a\left(1 - e^{-b_1 t}\right) & ; \ 0 \le t \le \eta_1 \\ a\left[1 - e^{-b_1 \eta_1 - b_2 (t - \eta_1)}\right] & ; \ \eta_1 < t \le \eta_2 \\ a\left[1 - e^{-b_1 \eta_1 - b_2 (\eta_2 - \eta_1) \ldots . b_{k+1} (t - \eta_k)}\right]; & t \ge \eta_k \end{cases}$$

The log-likelihood function is.

$$\log L_k(\eta_1, \ldots, \eta_k, a, b_1, b_2, \ldots b_{k+1}) = \log a + \left\{a e^{-b_1 \eta_1 - b_2 (\eta_2 - \eta_1) \ldots b_{k+1} (t - \eta_k}\right\} + \sum_{i=1}^{N(\eta_1)} \left\{\log(a b_1^2) + \log(t) - b_1 t_i\right\} + \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots + \sum_{i=N(\eta_k+1)}^{n} \left\{\log(a b_{k+1}^2) + \log(t) \ldots\ldots b_1 \eta_1 - b_2 (\eta_2 - \eta_1) \ldots\ldots - b_{k+1} (t - \eta_k)\right\}$$

The change points are restricted in the interval $[t_1, t_n]$, the log-likelihood function is bounded and estimates of $\hat{\eta}_1, \hat{\eta}_2, \ldots, \hat{\eta}_k, \hat{b}_1, \hat{b}_2, \ldots, \hat{b}_{k+1}$ can be maximum log likelihood function is expressed as,

$$\log L_k\left(\hat{\eta}_1, \hat{\eta}_2, \ldots\ldots\ldots, \hat{\eta}_k, \hat{b}_1, \hat{b}_2, \ldots, \hat{b}_{k+1}\right) = \max_{\eta_1, \ldots, \eta_k \in (t_2, t_{n,})} \max_{a, b_1, \ldots, b_{k+1}} \log L_k\left(\eta_1, \eta_1, \ldots\ldots a, b_1, b_2, \ldots . b_{k+1}\right)$$

The estimates $\hat{\eta}_1, \hat{\eta}_2, \ldots, \hat{\eta}_k, \hat{b}_1, \hat{b}_2, \ldots, \hat{b}_{k+1}$ can be obtained similarly.

4. Test Statistics of Single Change Point

Suppose the failure times $t_1, t_2, \ldots, t_n$ are distributed as the order statistics in a independent and identical random sample of size n from the density,

$$f(t; b_1, b_2, \eta) = \frac{\lambda_1(t)}{m_1(t)} I(0 < t < T)$$

$$= \begin{cases} \dfrac{b_1^2 t e^{-b_1 t}}{(1 - e^{-b_1 \eta - b_2 (T - \eta)})} & ; \quad 0 \le t \le \eta \\ \dfrac{b_2^2 t e^{-b_2 t}}{(1 - e^{-b_1 \eta - b_2 (T - \eta)})} & ; \quad \eta < t < T \\ 0 & ; \quad t \ge t \end{cases}$$

Where $m(T) = \int_0^T \lambda(t)\, dt$ and $I(.)$ are the indication functions?

So we construct the likelihood ratio test statistics:

$$S_n = 2\log \frac{\max_{b_1,b_2,\eta\in[t_2,t_{n-1}]}\prod_{i=1}^{n} f(t_i;b_1;b_2,\eta)}{\max_{b_1=b_2,\eta\in[t_2,t_{n-1}]}\prod_{i=1}^{n} f(t_i;b_1;b_2,\eta)}$$

$$= 2\log \frac{\max_{\eta\in[t_2,t_{n-1}]}\max_{a,b_1,b_2} L_1(\eta,a,b_1,b_2)}{\max_{a,b} L_0(a,b)}$$

$$= 2\left[\log L_1(\hat{a},\hat{b}_1,\hat{b}_2,\hat{\eta}) - \log L_0(\tilde{a},\tilde{b})\right] \quad \ldots (18)$$

5. Steps for Testing Procedure

STEP: 1- For the S-Shaped delayed model the estimates of $\tilde{a},\tilde{b}$ are obtained by solving equation 4 and 5.

Step: 2- For the S-Shaped delayed model with change-point, the estimates $\hat{\eta},\hat{a},\hat{b}_1,\hat{b}_2$ can be obtained by comparing the values of $\log L_1(t_m,a_1,b_{1m},b_{2m}); m = 2,\ldots,n-1$, and $\text{Log}\,L_1(t_m,\bar{a}_m,\bar{b}_{1m},\bar{b}_{2m})$ as discussed earlier.

Step: 3- Calculate S_n by equation (18).

Step: 4- The test at α-level is to reject H_0 if

$S_n > \chi_1^2(\alpha)$,

Where $\chi_1^2(\alpha)$ is the upper α point of the χ^2-distribution with one degree of freedom. Otherwise we accept the hypothesis.

5.1 An Example

The data set shown in table 1. Are collected from the system ***T1*** and ***Musa*** (1979). This data set includes 136 faults in the testing phase. Here we use the method proposed earlier to test the existence of change-points. The S-Shaped Delayed model fit of this data with one change-point resulted in parameter estimates of $\hat{a}_0 = 140.3$ and $\hat{b}_0 = 3.75\times10^{-5}$.

The S-Shaped Delayed model with one change-point resulted in parameter estimates of $\hat{a}_1 = 146.4$; $\hat{b}_{11} = 1.76\times10^{-4}$; $\hat{b}_{12} = 3.12\times10^{-5}$, and $\hat{\tau} = 1058$ (i.e., m = 16). Our estimation of change-point agrees with that of Wang and Wang (2005) and Zou (2003).

Although it is clear that the estimates $\hat{b}_{11}$ and $\hat{b}_{12}$ are significantly different. If the difference between the estimates $\hat{b}_{11}$ and $\hat{b}_{12}$ is larger than the threshold value, then there is a change-point. Here we can use out test statistics to test if $\hat{\tau}$ = 1056 is a change-point. The value of the test statistics is $S_n = 18.20 > \chi_1^2(0.05) = 3.84$, and we reject the null hypothesis i.e. there is a change-point in the testing process.

Table 1. Software failure times data: system T1

Software Failure times (CPUs)							
3	1846	5324	1025	1580	2677	4229	5648
33	1872	5389	1049	1618	2775	4229	5656
146	1986	5565	1062	1622	2846	4540	5702
227	2311	5623	1098	1635	2849	4665	6255
342	2366	6080	1111	1716	2936	4759	6265
351	2608	6380	1141	1745	3008	4829	6266
353	2676	6477	1144	1775	3240	4917	6373
444	3098	6740	1181	1828	3533	4941	6410
556	3278	7192	1255	1856	3679	5014	6489
571	3288	7447	1255	1872	3764	5204	7104
709	4434	7644	1279	1955	3765	5248	7436
759	5034	7837	1312	2056	3791	5287	7540
836	5049	7843	1348	2101	3971	5332	7605
860	5085	7922	1470	2130	4058	5344	8154
968	5089	8738	1525	2306	4201	5443	8270
105	5089	1008	1526	2412	4204	5538	8456
172	5097	1023	1527	2591	4218	5646	8868

6. Conclusion

In software reliability the problem of change-point is considered and some NHPP software reliability model with change-point has been proposed. Practically, the change-point is unknown, and it is possible that there is more than one change-point. In this article, we construct test statistics to test the existence of change-point by using S-Shaped Model. In the testing process we find that there is existence of change-points.

In the software testing phase, sometimes the failure cannot be observed exactly and only the number of failures up to a given time is known. The data use of this testing is a grouped data. But, the limitation of my study

that my test statistics is not used for the grouped data. Chang (2001) suggested the testing for grouped type of data.

References

(1) Chang, Y.P. (2001) : Estimation of parameters for non-homogeneous Poisson process: software reliability with change point model, Communication in Statistics: Simulation and Computation; 30; 623-635.

(2) Chen, J., Gupta, A. K. (2001): On change-point detection and estimation; Communication in Statistics: Simulation and Computation; 30; 665-697.

(3) Goel, A. L., Okumoto, K. (1979) : Time dependent error detection rate model for software reliability and other performance measures, IEEE Transaction on Reliability;28;206-211.

(4) Jeliski, Z., Moranda, P.B. (1972) : Software reliability research, In : Freiberger, W., ed. Statistics computer Performance Evaluation. New York: Academic Press, pp. 465-497.

(5) Nguyen, H. T., Rogers, G.S., Walker, E. A. (1984): Estimation in change-point hazard rate models; Biometrika; 71;299-304.

(6) Zhao. (1993): Change-point problem in software and hardware reliability, Communication in Statistics: Theory and Methods; 22; 757-768.

(7) Zou, F. Z. (2003) : change-point perspective on the software failure process; Software Testing Verification and Reliability;13;85-93.

A Comparative Study of Improved Region Selection Process in Image Compression using SPIHT and WDR

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Abstract: The SPIHT algorithm was powerful, efficient and simple image compression algorithm. By using this algorithm, the highest PSNR values for given compression ratios for a variety of images can be obtained. SPIHT stands for Set Partitioning in Hierarchical Trees. SPIHT was designed for optimal progressive transmission, as well as for compression. The important SPIHT feature is its use of embedded coding. The pixels of the original image can be transformed to wavelet coefficients by using wavelet filters. The problem in SPIHT is that it only implicitly locates the position of significant coefficients. This makes it difficult to perform operations, such as region selection on compressed data. By region selection, selecting a portion of a compressed image which requires increased resolution. Compressed data operations are possible with the Wavelet Difference Reduction (WDR) algorithm. The term difference reduction refers to the way in which WDR encodes the locations of significant wavelet transform values, in this paper I describe about SPIHT and WDR both are compressed algorithm with embedded coding. By experimental approach, I find that WDR is better than SPIHT for high resolution images.

***Keywords*:** image compression, Set Partitioning in Hierarchical Trees, significant and insignificant pixels, Wavelet Difference Reduction, bit plane encoding.

1. Compression using SPIHT

One of the defects of SPIHT[1] is that it only implicitly locates the position of significant coefficients. This makes it difficult to perform operations, such as region selection on compressed data, which depend on the exact position of significant transform values. By region selection, also known as region of interest (ROI), we mean selecting a portion of a compressed image which requires increased resolution. The term difference reduction refers to the way in which WDR encodes the locations of significant wavelet transform values. WDR[4] can produce perceptually superior images, especially at high compression ratios.

In a progressive transmission method, the decoder starts by setting the reconstruction image to zero. It then inputs (encoded) transform coefficients, decodes them, and uses them to generate an improved reconstruction image. The main aim in progressive transmission is to transmit the most important image information first. This is the information that results in the largest reduction of the distortion. SPIHT uses the mean squared error (MSE) distortion measure.

$$D_{mse}(P - \hat{P}) = \frac{|P-\hat{P}|^2}{N} = \frac{1}{N}\sum_I \sum_J (P_{IJ} - \hat{P}_{IJ})^2 \quad \ldots (1)$$

Where N is the total number of pixels. So the largest coefficients contain the information that reduces the MSE distortion.

1.1 SPIHT Process

SPIHT sorts the coefficients and transmits their most significant bits first. A wavelet transform has already been applied to the image and that the transformed coefficients are sorted.

The next step of the encoder is the refinement pass. The encoder performs a sorting step and a refinement step in each iteration. SPIHT uses the fact that sorting is done by comparing two elements at a time, and each comparison results in a simple yes/no result. The encoder and decoder use the same sorting algorithm, the encoder can simply send the decoder the sequence of yes/no results, and the decoder can use those to duplicate the operations of the encoder. The main task of the sorting pass in each iteration is to select those coefficients that satisfy 2n<=|ci,j |<2n+1. This task is divided into two parts. For a given value of n, if a coefficient ci,j satisfies |ci,j|>=2n, then that it is said as significant; otherwise, it is called insignificant. The encoder partitions all the coefficients into a number of sets Tk and performs the significance test.

$$S_n(T) = \begin{cases} 1, \max_{(i,j)\in T} |C_{I,J}| \geq 2^N \\ 0, Otherwise. \end{cases}$$

… (1.2)

On each set Tk. The result may be either "no" This result is transmitted to the decoder. If the result is "yes," then Tk is partitioned by both encoder and decoder, using the same rule, into subsets and the same significance test is performed on all the subsets. This partitioning is repeated until all the significant sets are reduced to size 1. The result, Sn(T), is a single bit that is transmitted to the decoder.

The sets Tk are created and partitioned using a spatial orientation tree. This set partitioning sorting algorithm uses the following four sets of coordinates:

1. The set contain the coordinates of the four offspring of node is Off [i,j]. If node is a leaf of a spatial orientation tree, then Off [i, j] is empty.
2. The set contain the set of coordinates of the descendants of node is called Des[i,j].
3. The set contain the set of coordinates of the roots of all the spatial orientation trees called R.
4. Next the set is a difference set Des[i,j]- Off[i,j]. This set contains all the descendants of tree node except its four offspring as Diff [i,j].

The spatial orientation trees are used to create and partition the sets Tk. The partitioning rules are given below:

1. Each spatial orientation tree need initial set.
2. If set Des[i, j] is significant, then it is partitioned into Diff[i, j] plus the four single element sets with the four offspring of the node.
3. If Diff[i, j] is significant, then it is partitioned into the four sets Des[k, l], where k=1..4 of node .The Fig. 3.1 shows the Spatial Orientation Trees in SPIHT.

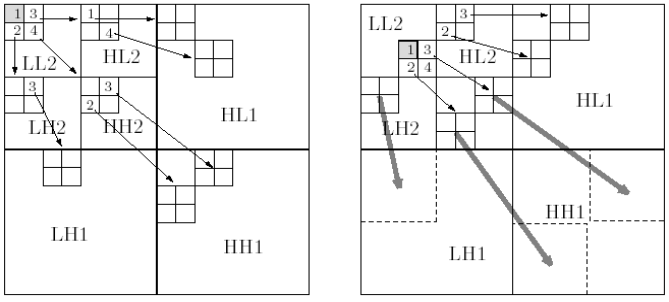


Figure 3.1: Spatial Orientation Trees in SPIHT

1.2 SPIHT Algorithm

It is important to have the encoder and decoder test sets for significance .So the coding algorithm uses three lists called SP for list of significant pixels , initialized as empty, IP is list of insignificant pixels for the coordinates of all the root node belongs to root set R, and IS is list of insignificant sets to the coordinates of all the root node in R that have descendants and treated as special type entries.

Procedure:

Step 1: Initialization: Set n to target bit rate.

```
for each node in IP do:
  if Sn [ i, j] = 1,(according to eq 1.2)
   move pixel coordinates to the SP and
    keep the sign of ci,j ;
    X: for each entry in the IS do the following steps:
              if the entry is root node with descendants
               if Sn(Des[i, j]) = 1, then
                  for each offspring (k, i ) in Off[i, j] do:
                           if ( Sn(k, i) = 1) then
                                { add to the SP,
          output the sign of ck,l;}
      else
       attach (k, l) to the IP;
        if (Diff[i, j] <> 0)
          {move (i, j) to the end of the IS,
          go to X;}
  else
    remove entry from the IS;
  If the entry is root node without descendants then
       output Sn(Diff[i, j]);
       if Sn(Diff[i, j]) = 1, then
       append each (k, l) in Off(i, j) to the IS as a special
       entry and remove node from the IS:
```

Step 3: Refinement pass: for each entry in the SP, except those included in the last process for sorting , output the nth most significant bit of |i,j|;

Step 4: Loop: reduced n by 1 and go to X if needed.

2. Compression using WDR

Although WDR can produce perceptually superior images, especially at high compression ratios. The WDR compression and decompression systems are shown in Fig. 2.1 and Fig. 2..2.

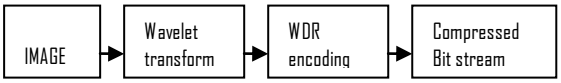


Figure 2.1: WDR Compression Steps

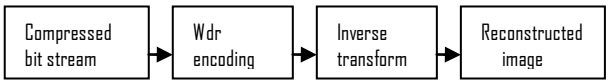


Figure 2.2:WDR Decompression Steps

The only difference between WDR and the Bit-plane encoding is in the significance pass. In WDR, the output from the significance pass consists of the signs of significant values along with sequences of bits which concisely describe the precise locations of significant values.

2.1 WDR Algorithm

The WDR algorithm is a very simple procedure. A wavelet transform is first applied to the image, and then the bit-plane based WDR encoding algorithm for the wavelet coefficients is carried out. WDR mainly consists of five steps as follows:

Step 1: Initialization: During this step an assignment of a scan order should first be made. For an image with P pixels, a scan order is a one-to-one and onto mapping $\tilde{F}_{i,j}$ = Xk , for k =1,2,..., P between the wavelet coefficient ($\tilde{F}_{i,j}$) and a linear ordering (Xk). The scan order is a zigzag through sub bands from higher to lower levels. For coefficients in sub bands, row-based scanning is used in the horizontal sub bands, column based scanning is used in the vertical sub bands, and zigzag scanning is used for the diagonal and low-pass sub bands. As the scanning order is made, an initial threshold T0 is chosen so that all the transform values satisfy |Xm|< T0 and at least one transform value satisfies |Xm|>= T0 / 2.

Step 2: Update threshold: Let Tk=Tk-1 / 2.

Step 3: Significance pass: In this part, transform values are significant if they are greater than or equal to the threshold value. The difference reduction method consists of a binary encoding of the number of steps to go from the index of the last significant value to the index of the current significant value. The output from the significance pass is the signs of significant values along with sequences of bits, generated by difference reduction, which describes the precise locations of significant values.

Step 4: Refinement pass: The refinement pass is to generate the refined bits via the standard bit-plane quantization procedure like the refinement process in SPHIT method. Each refined value is a better approximation of an exact transform value.

Step 5: Repeat steps (2) through (4) until the bit budget is reached.

3. Experiment with SPIHT

3.1 Experimental Images

The images Lena, Cameraman and Boat are used for the experiments. The original images are shown in Fig. 3.1(a), Fig. 3.2(a), and Fig. 3.3(a), The results of experiments are used to find the PSNR (Peak Signal to Noise Ratio) values using the formulae

$$PSNR = 10 log_{10}\left(\frac{255^2}{\sqrt{mse}}\right) dB$$

Where P is source image $\hat{p}$ is image reconstructed by compression, N total no of pixels is range of pixel values. PSNR is the ratio between the maximum possible power of a signal and power of corrupting noise that affect the fidelity of its representation. It expressed by logarithmic decibel scale. It is used to measure the quality of reconstruction of compression codec's. If PSNR is high , then reconstruction of image quality is high. The formulae for finding MSE (Mean Square Error) values for the reconstructed image are given. here P and $\hat{P}$ are noisy approximation of one another. When two images are identical MSE value is zero.

$$MSE = \frac{\sum [P_{IJ} - \hat{P}_{IJ}]^2}{N}$$

In SPIHT, a wavelet transform is applied to the input image and the wavelet coefficients are got. Then the wavelet coefficients are sorted. Then in each iteration a sorting step and a refinement step can be performed. Embedded coding method is used to encode the coefficients. Then the decoder starts by setting the reconstruction image to zero. It then gets the transform coefficients, decodes them, and generates an improved reconstruction image. The results that got by using SPIHT technique are shown in the Fig. 3.1(b), Fig. 3.2(b), Fig. 3.3(b). Some of the best results highest PSNR values for given compression ratios for the sample images have obtained with SPIHT.


Figure 3.1: (a)
Lena Original Image


3.1(b)
Compressed by SPIHT


Figure 3.2: (a)
Original Image


3.2 (b)
After Compression using SPIHT


Figure 3..3: (a)
Original Image


3.3(b)
After Compression using SPIHT

4. Experiment with WDR

4.1 Experimental Images

The images Lena, Cameraman, and Boat are used for the experiments. The original images are shown in Fig. 4.1(a), Fig. 4.2(a) and Fig. 4.3(a), The results of experiments are used to find the PSNR (Peak Signal to Noise Ratio) values and MSE (Mean Square Error) values for the reconstructed images. WDR employs similar encoding stages to SPIHT.

It also conducts a sorting pass and a refinement pass for each bit plane. As the counterparts of the three lists in SPIHT, three sets are defined in WDR, i.e. the set of insignificant coefficients (ICS), the set of significant coefficients (SCS), and the temporary set of significant coefficients (TPS). Since WDR does not utilize the zero tree data structure, it does not have a list of insignificant sets as in SPIHT. Instead of directly concatenating the significant coefficients found in a given bit plane to the SCS, like SPIHT adding such coefficients to the LSP, WDR adds newly identified significant coefficients to the TPS. The TPS is later concatenated to the end of the SCS after the refinement pass. The only difference in bit plane encoding between WDR and SPIHT is in the sorting pass. Instead of using zero trees to represent insignificant coefficients, WDR defines a scan order of wavelet coefficients, which traverses all sub bands in a wavelet pyramid from coarse resolutions to fine resolutions. The results that got by using WDR technique are shown in the Fig. 4.1(b), Fig. 4.2(b) and Fig. 4.3(b). Some of the best results highest PSNR values for given compression ratios for the sample images have obtained with WDR.


Figure 4.1: (a)
Original Image


4.1 (b)
Compressed by WDR Compression


Figure 4.2: (a)
Original Image


4.2(b)
Compressed by WDR


Figure 4.3: (a)
Original Image


4.3(b)
Compressed by WDR

5. Performance Analysis

The PSNR values for the images compressed by SPIHT and WDR are tabulated in Table 4.1. The MSE values for the images compressed by SPIHT and WDR are tabulated in Table 4.2. The graphical representation of PSNR and MSE values are expressed as a bar graph are shown in Fig. 4.3 and Fig. 4.4.

TABLE 4.1

PSNR VALUES FOR SPIHT Vs WDR COMPRESSION

Image	SPIHT	WDR
Lena	39.85	32.34
Cameraman	35.56	26.07
Boat	37.59	28.85

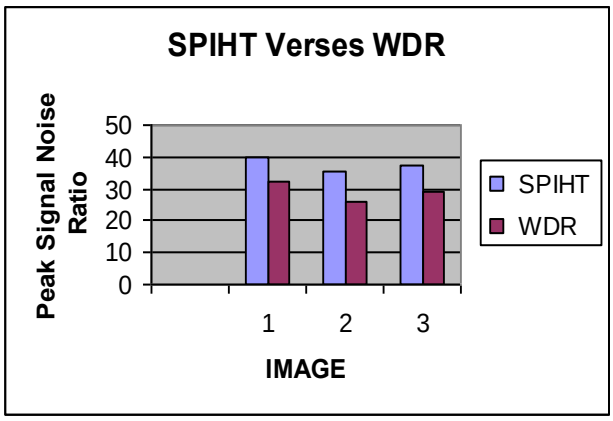
TABLE 4.2

MSE VALUES FOR SPIHT Vs WDR COMPRESSION

Image	SPIHT	WDR
Lena	6.7242	6.1524
Cameraman	18.0679	12.6651
Boat	11.32	9.20

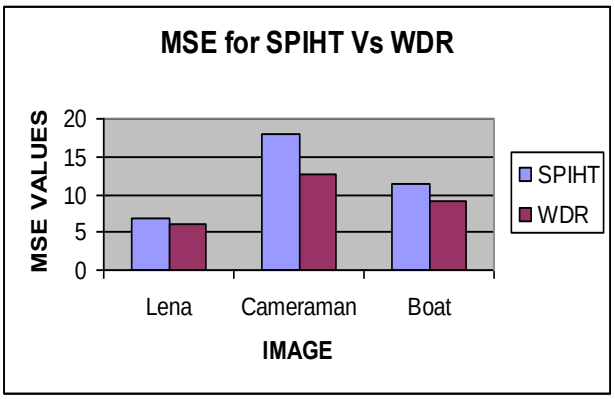
GRAPH 4.3

MSE values for SPIHT and WDR Compression



The SPIHT method provides highest image quality, progressive image transmission, fully embedded coded file, Simple quantization algorithm, fast coding/decoding, completely adaptive, lossless compression, exact bit rate coding and Error protection.

GRAPH 4.4



6. Conclusion

Furthermore, its embedded coding process proved to be effective in a broad range of reconstruction qualities. From the experiment and performance analysis it was observed that the reconstructed images are having high PSNR values and low MSE values.

It is not hard to see that WDR is of no greater computational complexity than SPIHT. For one thing, WDR does not need to search through quad trees as SPIHT does. The calculations of the reduced binary expansions add some complexity to WDR, but they can be done rapidly with bit-shift operations. WDR provides such good results when compare to SPIHT. From the experiment and performance analysis, it was observed that the PSNR and MSE values are good on the images.

Reference

[1] A. Said, W.A. Pearlman. “A new, fast, and efficient image codec based on set partitioning in hierarchical trees”. IEEE Trans. on Circuits and Systems for Video Technology, Vol. 6, No. 3, pp. 243-250, 1996.

[2] Shapiro J.M. “Embedded image coding using zero trees of wavelet coefficients”. IEEE Trans. Signal Proc., Vol. 41, No. 12, pp. 3445{3462, 1993.

[3] J. Tian, R.O. Wells. “A lossy image codec based in index coding”, IEEE Data Compression Conference, DCC’96, 1996, pp.456.

[4] J. Tian, R.O. Wells, Jr. “Image data processing in the compressed wavelet domain”. 3rd International Conference on Signal Processing Proc., B. Yuan and X. Tang, Eds., pp. 978{981, Beijing, China, 1996.

[5] Y. Yuan, M. K. Mandal. “Novel embedded image coding algorithms based on wavelet difference reduction”, in: Proceedings of IEEE International Conference on Vision, Image and Signal Processing, vol.152, 2005, pp. 9-19.

[6] Ahmed, N., Natarajan, T., and Rao, K. R. “Discrete Cosine Transform”, IEEE Trans. Computers, vol. C-23, Jan. 1974, pp. 90-93.

[7] M. Antonini, M. Bar laud, P. Mathieu, I. Daubechies. “Image coding using wavelet transform”. IEEE Trans. Image Proc., Vol. 5, No. 1, pp. 205-220, 1992.

[8] Anil K. Jain, Fundamentals of Digital Image Processing, Englewood Cliff, NJ: Prentice Hall, 1989.

[8] G.M. Davis, A. Nosratinia. “Wavelet-based Image Coding: An Overview. Applied and Computational Control”, Signals and Circuits, Vol. 1, No. 1, 1998.

Novel Intelligent Low Cost Child Disease Diagnostic System

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Abstract: In India, 30% to 40 % babies are low birth weight babies (LBW) as opposed to about 5% to 7% of newborn in the west. In India, 7 to 10 million LBW infants are born annually. About 10 % to 12% of Indian babies are born preterm (less than 37 completed weeks) as compared with 5% to 7% incidence in the west. These infants are physically immature and therefore their neonatal mortality is high. It is possible to increase the survival of the infants and quality of human life through prompt and adequate disease management of the newborn.

The proposed novel model of intelligent low cost child disease diagnostic system based on *Artificial Intelligence Algorithm* is helpful for diagnostic- cum- preventive approach to reduce the immaturity, fragility, vulnerability and dependence of the neonates in the developing countries like India especially in the state of M.P. , Bihar, U.P. and eastern states to reduce neonatal and child mortality. Secondly, a significant proportion of the pediatricians' time, especially in major hospitals, large cities and overpopulated areas is spent on examination and evaluation of apparently healthy babies and detection of minor developmental defects.

In addition to these facts, India and other third world countries face the major problem of child health diagnosis and malnutrition mostly in rural and remote part of it. Medical facilities and expertise is either absent or out of reach of these tribal and poor communities, many public health centers (PHCs) lack in advice by experts on immediate basis in case of emergencies. Major hurdles include the lack of medical experts and trained manpower, scarcity of funds and improper budgetary allocation for rural health at state and central government level.

Keywords—Artificial Neural Network, Infant Disease Management, Malaria, Typhoid, FFANN.

1. Introduction

Conventional medical diagnosis in clinical examinations relies highly upon the physicians' experience. Physicians intuitively exercise knowledge obtained from previous patients' symptoms. In everyday practice, the amount of medical knowledge grows steadily, such that it may become difficult for physicians to keep up with all the essential information gained. To quickly and accurately diagnose a patient, there is a critical need in employing computerized technologies to assist in medical diagnosis and access the related information. Computer-assisted technology is certainly helpful for inexperienced physicians in making medical diagnosis as well as for experienced physicians in supporting complex decisions.

Computer-assisted technology has become an attractive tool to help physicians in retrieving the medical information as well as in making decisions in face of today's medical complications. Machine learning techniques with computer-aided medical diagnosis should have good comprehensibility, i.e., the transparency of diagnostic knowledge and the explanation ability. Nowadays, as the computational power increases, the role of automatic visual inspection becomes more important. Image processing and artificial intelligence techniques are introduced that may provide a valuable tool. Currently in Malaysia the traditional method for the identification of Malaria parasites requires a trained technologist to manually examine and detect the number of the parasites subsequently by reading the slides.

This is a very time consuming process, causes operator fatigue and is prone to human errors and inconsistency. An automated system is therefore needed to complete as much work as possible for the identification of Malaria parasites. The integrated system including soft computing tools has been successfully designed with the capability to improve the quality of the image, analyze and classify the image as well as calculating the number of Malaria parasites [1]. The cost of such system is high enough and hence not recommended for third world countries, especially India which has a dense rural population living below poverty line. Such systems do not consider multiple diseases at the same time for diagnosis.

Medical expert systems in various areas are certain to grow because huge medical data are provided according to increment of performance of medical systems/scanners. In them, the most famous medical expert system would be MYCIN [2]. MYCIN did not use fuzzy logic directly; one of primary components was the use of certainty factors. The medical expert systems using fuzzy logic directly are in References [3], [4] and [5]. In them, fuzzy degree of uncertainty or possibility degree of certain diagnosis is employed.

Artificial Intelligence (AI) is the study of mental facilities through the use of computational models. It has produced a number of tools. These tools are of great practical significance in engineering to solve various complex problems normally requiring human intelligence. In general, an artificial neural network is built in two steps, that is, generating component artificial neural networks and then combining their predictions. The powerful tools among these are expert system (knowledge-based system), ANN, Genetic Algorithm based ANN, Neural-Fuzzy, and Support Vector Machines (SVM).

1.1 Expert System:

The expert system (ES), also known as knowledge-based systems (KBS), is basically computer programs embodying knowledge about a narrow domain for the solution of problems related to that domain. An ES mainly consists of a knowledge base and an inference mechanism. The knowledge base contains domain knowledge, which may be expressed as any combinations of 'IF-THEN' rules, factual statements, frames, objects, procedures and cases. The inference mechanism manipulates the stored knowledge to

produce solutions. The general structure of a knowledge based system is given in the Fig.1.

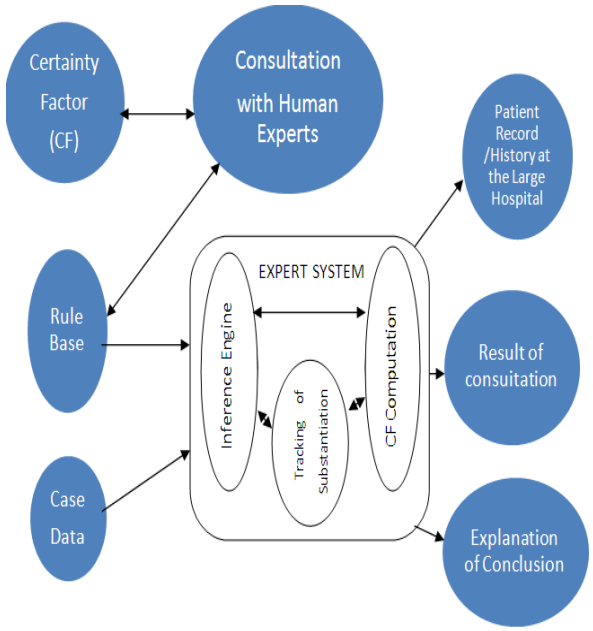

Fig. 1 General Structure of a Knowledge Based System

1.2 Fuzzy Logic System:

A demerit of an ordinary rule-based ES is that they cannot handle new situations not covered explicitly in their knowledge bases. Hence, ESs cannot give any conclusions in these situations. The fuzzy logic systems (FLSs) are based on a set of rules. These rules allow the input to be fuzzy, i.e. more like the natural way that human express knowledge [XS]. The use of fuzzy logic can enable ESs to be more practical. The knowledge in an ES employing fuzzy logic can be expressed as fuzzy rules (or qualitative statements). A reasoning procedure, the compositional rule of inference and conclusions are to be drawn by extrapolation or interpolation from the qualitative information stored in the knowledge base. The Overall decision support system is shown in Fig. 2.

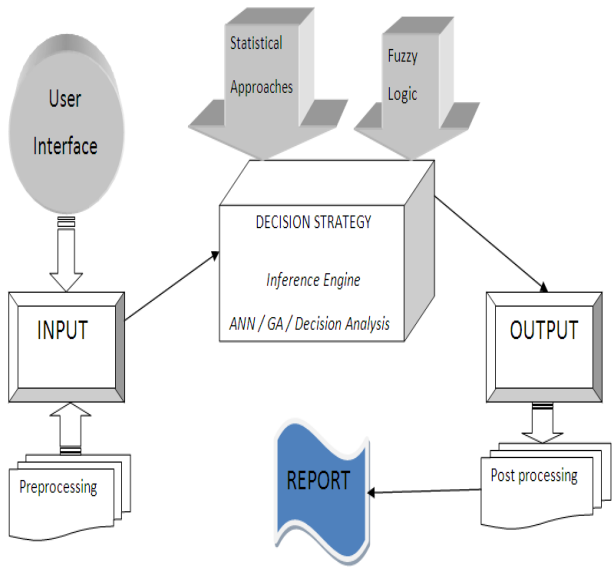

Fig.2 Decision-support System

1.3 Artificial Neural Network:

Artificial neural network (ANN) can capture domain knowledge from examples they can readily handle both continuous and discrete data and have good generalization capability as with fuzzy expert systems. An ANN is a computational model of the human brain. ANNs assume that computation is distributed over several simple units called neurons, which are interconnected and operate in parallel thus known as parallel distributed processing systems or connectionist systems. Implicit knowledge is built into a neural network by training it. Some ANNs can be trained by typical input patterns and the corresponding expected output patterns. The error between the actual and expected outputs is used to strengthen the weights of the connections between the neurons. This type of training is known as supervised training. Some of the ANNs are trained in an unsupervised mode, where only the input patterns are provided during training and the network learns automatically to cluster them in groups with similar features.

1.4 Genetic Algorithm:

Genetic algorithm (GA) is a stochastic optimization procedure inspired by natural evolution. It can yield the global optimum solution in a complex multi-model search space without requiring specific knowledge about the problem to be solved. A genetic or evolutionary algorithm operates on a group or population of chromosomes at a time, iteratively applying genetically based operators such as crossover and mutation to produce fitter populations containing better solution chromosomes.

1.5 Support Vector Machine:

Support Vector Machines (SVMs) are the methods for creating functions from a set of labeled training data. The function can be a classification function or the function can be a general regression function. For classification, SVMs operate by finding a hyper surface in the space of possible inputs, which will attempt to split the positive examples from the negative examples.

In this paper a novel intelligent very low cost child disease diagnostic system using ANN is proposed which helps in diagnostic-cum-preventive approach to reduce the immaturity, fragility, vulnerability and dependence of the neonates.

There are a number of different answers possible to the question of how to define neural networks. At one extreme, the answer could be that neural networks are simply a class of mathematical algorithms, since a network can be regarded essentially as a graphic notation for a large, class of algorithms. Such algorithms produce solutions to a number of specific problems. At the other end the reply may be that these are synthetic networks that emulate the biological neural networks found in living organism. In light of today's limited knowledge of biological neural networks and organisms, the more plausible answer seems to be closer to the algorithmic one.

There has been a long history of interest in the biological sciences on the part of engineers, mathematicians, and physicists endeavoring to gain new ideas, inspirations, and designs. Artificial neural networks have undoubtedly been

biologically inspired, but the close correspondence between them and real neural systems is still rather weak. Vast discrepancies exist between both the architectures and capabilities of artificial and natural neural networks. Knowledge about actual brain functions is so limited, however, that there is little to guide those who try to emulate it.

Despite of loose analogy between artificial and natural neural system, we will briefly review the biological neuron model.

2. Biological Neuron

The elementary nerve cell, called a neuron, is the fundamental building block of the biological neural network. Its schematic diagram is shown in fig. A typical cell has three major regions: the cell body, which is also called the soma, the axon, and the dendrites. Dendrites form the dendritic tree, which is a very fine bush of thin fibers around the neuron's body. Dendrites receive information from neurons through axons-long fibers that serve as transmission lines. An axon is a long cylindrical connection that carries impulses fron the neuron. The end part of an axon splits into a fine arborization. Each branch of it terminates in a small endbulb almost touching the dendrites of neighboring neuron. The axon-dendrite contact organ is called a synapse. The synapse is where the neuron introduces its signal to the neighboring neuron. The signals reaching a synapse and received by dendrites are electrical impulse. The inteneuronal transmission is sometimes electrical but is usually effected by the release of chemical transmitters at the synapse.

The neuron is able to respond to the total of its inputs aggregated within a short time interval called the period of latent summation. The neuron's response is generated if the total potential of its membrane reaches a certain level. Let us consider the conditions necessary for the firing of a neuron. Incoming impulses can be excitatory if they cause the firing, or inhibitory if they hinder the firing of the response. A more precise condition for firing is that the excitation should exceed the inhibition by the amount called the threshold of the neuron. Since a synaptic connection causes the excitatory or inhibitory reactions of the receiving neuron, it is practical to assign positive and negative unity weight values, respectively, to such connections. This allows us to reformulate the neuron's firing condition. The neuron fires when the total of the weights to receive impulses exceeds the threshold valve during the latent summation period.

3.Details of the artificial intelligent method employed

ANNs are highly interconnected processing units inspired in the human brain and its actual learning process. Interconnections between units have weights that multiply the values which go through them. Also, units normally have a fixed input called bias. Each of these units forms a weighted sum of its inputs, to which the bias is added. This sum is then passed through a transfer function.

Prediction with ANNs involves two steps, one is *training* and the other is *learning*. Training of Feed forward artificial neural networks (FFANNs) is normally performed in a supervised manner. The success of training is greatly affected by proper selection of inputs. In the learning process, a neural network constructs an input-output mapping, adjusting the weights and biases at each iteration based on the minimization or optimization of some error measured between the output produced and the desired output. This process is repeated until an acceptable criterion for convergence is reached. The most common learning algorithm is the back propagation (BP) algorithm, in which the input is passed layer through layer until the final output is calculated, and it is compared to the real output to find the error. The error is then propagated back to the input adjusting the weights and biases in each layer. The standard BP learning algorithm is a steepest descent algorithm that minimizes the sum of square errors. In order to accelerate the learning process, two parameters of the BP algorithm are adjusted: the learning rate and the momentum. The learning rate is the proportion of error gradient by which the weights are to be adjusted. Larger values give a faster convergence to the minimum. The momentum determines the proportion of the change of past weights that are used in the calculation of the new weights.

In this paper, the fully-connected feed forward multilayer perceptron network is used and trained. The network consists of an input layer representing the input data to the network, hidden layers and an output layer representing the response of the network. Each layer consists of a certain number of neurons; each neuron is connected to other neurons of the previous layer through adaptable synaptic weights *w* and biases *b*.

If the inputs of neuron j are the variables x_1, x_2, .. x_i, .., x_N, the output u_j of neuron *j* is obtained as ,

$$u_j = \varphi\left(\sum_{i=1}^{N} w_{ij} x_i + b_j\right)$$

where, w_{ij} is the weight of the connection between neuron *j* and *i*-th input; b_j is the bias of neuron *j* and φ is the transfer (activation) function of neuron *j*.

An ANN of three layers (one hidden layer) is considered with N, M and Q neurons for the input, hidden and output layers, respectively. The input patterns of the ANN represented by a vector of variables $x = (x_1, x_2, \ldots, x_i, \ldots, x_N)$ submitted to the NN by the input layer are transferred to the hidden layer. Using the weight of the connection between the input and the hidden layer and the bias of the hidden layer, the output vector $u = (u_1, u_2, \ldots, u_j, \ldots, u_M)$ of the hidden layer is determined.

The output u_j of neuron j is obtained as,

$$u_j = \varphi hid\left(\sum_{i=1}^{N} w_{ij}^{hid} x_i + b_j^{hid}\right)$$

where, w_{ij}^{hid} is the weight of connection between neuron *j* in the hidden layer and the *i*-th neuron of the input layer, b_j^{hid} represents the bias of neuron *j* and φhid is the activation function of the hidden layer.

The values of the vector *u* of the hidden layer are transferred to the output layer. Using the weight of the connection between the hidden and output layers and the bias of the output layer, the output vector $y = (y_1, y_2, \ldots y_k, \ldots, y_Q)$ of the output layer is determined.

The output y_k of neuron k (of the output layer) is obtained as,

$$y_k = \varphi out\left(\sum_{j=1}^{M} w_{jk}^{out}\, u_j + b_k^{out}\right)$$

where, w_{jk}^{out} is the weight of the connection between neuron k in the output layer and the j-th neuron of the hidden layer, b_k^{out} is the bias of neuron k and φout is the activation function of the output layer.

The output y_k is compared with the desired output (target value) y_k^d . The error E in the output layer between y_k and y_k^d ($y_k^d - y_k$) is minimized using the mean square error at the output layer (which is composed of Q output neurons), defined by,

$$E = \frac{1}{2}\sum_{k=1}^{Q}(y_k^d - y_k)^2$$

Training is the process of adjusting connection weights w and biases b. In the first step, the network outputs and the difference between the actual (obtained) output and the desired (target) output (i.e., the error) is calculated for the initialized weights and biases (arbitrary values). In the second stage, the initialized weights in all links and biases in all neurons are adjusted to minimize the error by propagating the error backwards (the BP algorithm). The network outputs and the error are calculated again with the adapted weights and biases, and this training process is repeated at each epoch until a satisfied output y_k is obtained corresponding with minimum error. This is done by adjusting the weights and biases of the BP algorithm to minimize the total mean square error and is computed as,

$$\Delta w = w^{new} - w^{old} = -\eta \frac{\partial E}{\partial w} \qquad (1a)$$

$$\Delta b = b^{new} - b^{old} = -\eta \frac{\partial E}{\partial b} \qquad (1b)$$

where, η is the learning rate. Equations (1a) and (1b) show the generic rule used by the BP algorithm. Equations (2a) and (2b) illustrate this generic rule of adjusting the weights and biases. For the output layer, we have,

$$\Delta_{wjk}^{new} = \alpha \Delta_{wjk}^{old} + \eta\, \delta_k\, y_k \qquad (2a)$$

$$\Delta_{bk}^{new} = \alpha \Delta_{bk}^{old} + \eta\, \delta_k \qquad (2b)$$

where, α is the momentum factor (a constant between 0 and 1) and $\delta k = y_k^d - y_k$

For the hidden layer, we get,

$$\Delta_{wij}^{new} = \alpha \Delta_{wij}^{old} + \eta\, \delta_j\, y_j \qquad (3a)$$

$$\Delta_{bj}^{new} = \alpha \Delta_{bj}^{old} + \eta\, \delta_j \qquad (3b)$$

where,

$$\delta_j = \Sigma_k^Q \delta_k\, w_{jk} \quad \text{and} \quad \delta_k = y_k^d - y_k$$

The general structure of a Supervised Learning Data-Based System is shown in Fig.3.

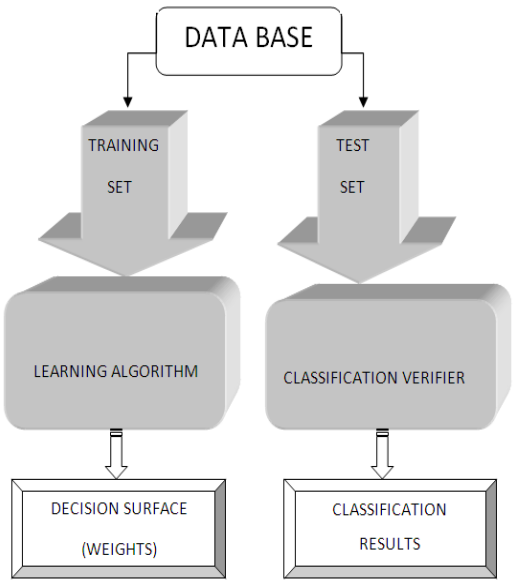

Fig. 3. General structure of Supervised Learning Data-Based System

4. Symptomatic studies of some diseases

4.1 Typhoid:

Typhoid is also known as enteric fever or salmonellosis. It is an infectious disease and is a very common cause of persistent high grade fever. Typhoid is a bacterial disease caused by *Salmonella typhi*. A related bacterium called *Salmonella paratyphi* causes paratyphoid fever. The disease is transmitted by contaminated food or water. These bacteria live within the gall bladders of some human beings without causing disease for years. These carriers pass these bacteria in their stools and if the carrier is a food handler, the disease spreads to a large number of people. The illness is also spread by a contaminated water supply. Since the bacteria are passed in the stools of the carriers as well as patients afflicted with acute illness, any contamination of the water supply with sewage spreads the disease in epidemic proportions [6].

4.2 The symptoms of the Typhoid:

Fever is the main symptom that gradually increases over four to five days. The fever is high grade (up to 40.5ºC or 105ºF) and almost continuous unless some fever-relieving drugs (antipyretics) are taken. The appetite is poor and the patient feels weak. The liver and spleen become enlarged. In serious conditions, perforation of the intestines may occur in a few cases[6].

4.3 Malaria:

Malaria is a parasitic disease characterized by high fever, chills and rigors. *Falciparum* malaria, one of four different types, affects a greater proportion of the red blood cells than the other types and is more serious. The disease is a major health problem in India as in most of the tropics and subtropics. Malaria is caused by a parasite (*Plasmodium*) that is transmitted from one human to another by the bite of infected anopheles mosquitoes. The symptoms occur in cycles of 48 to 72 hours. This is the time taken by the parasites to multiply inside the red blood cells, which then rupture, and the parasites infect more red blood cells. Malaria can also be transmitted congenitally (from a mother to her unborn baby) and, rarely, by blood transfusions [6].

4.4 Symptoms of Malaria:

There are sequential chills, fever, and sweating accompanied by headache, nausea and vomiting, muscle pain and anemia. In severe cases, there may also be jaundice, convulsions or coma [6].

Table 1 shows the sample data collection and weight assignment for malaria and typhoid. Data for normal patient with normal range is also given in the fourth column for immediate comparison.

Table 1:

Constraints	Malaria	Thyphoid	Normal
Onset	Sudden fever 0.6 to1.0	Slow rising fever .0.1 to 0.4	No fever/Small fever 0.1 to0.3
Kind of fever	Periodic 1	Continuous throughout day -1	Any type possible 1 or –1
Rose spots on body	Absent	Present 0.5 to 1	Absent 0.1 to 0.2
Heart beat relative to temperature	Increases with temperature 0.5 to 1	Decreases with temperature -0.5 to –1	Normal. No change with temperature 0.2 to –0.2
Leukopenia	Absent 0 to 0.2	Present 0.6 to 1	Absent 0 to 0.2
RBC Ring formation	Present 0.8 to 1	Absent 0 to 0.1	Absent 0 to 0.1
Tongue	Normal 0 to 0.1	Thickly coated V-Shaped 0.6 to 1	Normal 0 to 0.1
Spleen enlargement	No 0 to 0.2	Yes 0.5 to 0.8	No 0 to 0.2
Plasmodium test	Positive 1	Negative 0	Negative 0
Widal test	Negative 0	Positive 1	Negative 0

5. Results and Discussion

The FFANN is a widely accepted classifier. However, the success of FFANN to distinguish between Malaria and Thyphoid is strongly related to the success in the pre-processing of its input data. The inputs should contain lot of information in order for the network to properly classify the events. In this paper, three-layer FFANN is used and trained with a supervised learning algorithm called back propagation (BP). The FFANN consists of one input layer, one hidden layer and one output layer. The input layer consists of neurons: the inputs to these neurons are various symptoms for Malaria and Thyphoid like temperature, abdominal pain, pulse, vomiting, rashes, joint pain etc. The output layer consists of two neurons representing the Malaria and Thyphoid. With respect to the hidden layer, it is customary that the number of neurons in the hidden layer is done by trial and error. The same approach is used in the proposed algorithm. Symptoms of two diseases were used as an input to the network. For generalization, the randomized data is fed to the network and is trained for different hidden layers. Various training methods of Conjugate Gradient (CG) BP and Levenberg– Marquardt BP are used for training the network and average minimum MSE on training and testing data is obtained. For all training methods, it is assumed that learning rate LR = 0.8, momentum MM = 0.7, data used for training purpose TR = 10%, for cross validation CV = 20% and for testing purpose TS = 70%. With these assumptions, the variation of average MSE and percent accuracy of classification for both Malaria and Thyphoid with respect to the number of processing elements in the hidden layer is obtained.

Fig. 4 shows variation of percent of classification accuracy with respect to the number processing elements in the hidden layer. It is found that in Levenberg–Marquardt BP (‘trainlm’) method for four processing elements in the hidden layer the minimum MSE (0.00012) is obtained, which the lowest value is obtained by any method and 100% classification between Malaria and Thyphoid, which means there is a clear discrimination between Malaria and Thyphoid with this method. Hence, this network of Levenberg–Marquardt BP ‘trainlm’ for learning rate LR = 0.8, momentum MM = 0.7, training data TR = 10%, cross validation CV = 20% and testing data TS = 70% with 04 number of processing elements in the hidden layer is the best-suited network.

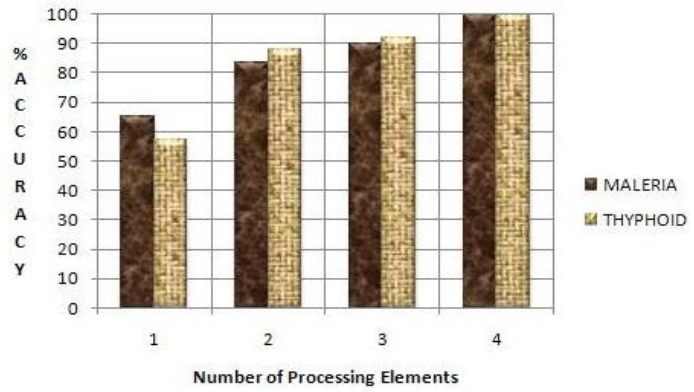

Fig.4 Variation of percentage accuracy with number of processing elements in hidden layer

6. Conclusion

The artificial neural network for the diagnosis provides an efficient way to assist doctor in the diagnosis of the Malaria and Typhoid. The feedback error propagation learning allows the programmer to use doctor's experience in training the network. The assistant of doctors can feed symptoms and the diagnosis given is similar to that of given by doctor itself.

Also it is a monotony free and presumption free diagnostic system. It also provides a better alternative to process abstract medical data over the conventional programming method.

References:

[1] S.F. Tohal and U.K. Ngah, *"Computer Aided Medical Diagnosis for the Identification of Malaria Parasites"*, IEEE - ICSCN 2007, MIT Campus, Anna University, Chennai, India. Feb. 22-24, 2007. pp.521-522.

[2] E. H. Shordiffe: *Computer-Based Medical Consultation, MYCIN.*, Elsevier /North Holland, New York ,1976.

[3] K. P. Adlassnig, *A survey on medical diagnosis and fuzzy subsets in M.M. Gupta and E. Sanchez (Eds): Approximate Reasoning in Decision Analysis*, North-Holland, pp.203-217, 1982.

[4] A.O. Esogbue, *"Measurement and valuation of a fuzzy mathematical model for medical diagnosis"* Fuzzy sets and Systems, 10, pp.223-242,1983.

[5] D. L. Hudson and M. E. Cohen, *"Fuzzy Logic in Medical Expert System"*, IEEE Eng. Med. and Bio., pp. 693-698, 1994.

[6] Suraj Gupte, *"The Short Textbook of Pediatrics"*, 11th Edition, Jaypee Pub. New Delhi, 2009.

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Tools and Techniques for Evaluating Web Information Retrieval Using Click-through Data

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Abstract. Search has arguably become the dominant paradigm for finding information on the World Wide Web. In order to build a successful search engine, there are a number of challenges that arise where techniques from artificial intelligence can be used to have a significant impact. In this paper, we explore a number of problems related to finding information on the web and discuss approaches that have been employed in various research programs, including some of those at Google. Specifically, we examine issues of such as web graph analysis, statistical methods for inferring meaning in text, and the retrieval and analysis of newsgroup postings, images, and sounds. We show that leveraging the vast amounts of data on web, it is possible to successfully address problems in innovative ways that vastly improve on standard, but often data impoverished, methods. We also present a number of open research problems to help spur further research in these areas.

Keywords: Information Retrieval, pageRank, Query, Search Engine

1. Introduction

Search engines are critically important to help users find relevant information on the World Wide Web. In order to best serve the needs of users, a search engine must find and filter the most relevant information matching a user's query, and then present that information in a manner that makes the information most readily palatable to the user. Moreover, the task of information retrieval and presentation must be done in a scalable fashion to serve the hundreds of millions of user queries that are issued every day to a popular web search engines such as Google. In addressing the problem of information retrieval on the web, there are a number of challenges in which Artificial Intelligence (AI) techniques can be successfully brought to bear. We outline some of these challenges in this paper and identify additional problems that may motivate future work in the AI research community.

We also describe some work in these areas that has been conducted at Google. We begin by briefly outlining some of the issues that arise in web information retrieval that showcase its differences with research traditionally done in Information Retrieval (IR), and then focus on more specific problems. Section 2 describes the unique properties of information retrieval on the web. Section 3 presents a statistical method for determining similarity in text motivated by both AI and IR methodologies. Section 4 deals with the retrieval of UseNet (newsgroups) postings, while Section 5 addresses the retrieval of non-textual objects such as images and sounds. Section 6 gives a brief overview of innovative applications that harness the vast amount of text available on the Web. Finally, Section 7 provides some concluding thoughts.

2. Information Retrieval on the Web

A critical goal of successful information retrieval on the web is to identify which pages are of high quality and relevance to a user's query. There are many aspects of web IR that differentiate it and make it somewhat more challenging than traditional problems exemplified by the TREC competition. Foremost, pages on the web contain links to other pages and by analyzing this web graph structure it is possible to determine a more global notion of page quality. Notable early successes in this area include the PageRank algorithm, which globally analyzes the entire web graph and provided the original basis for ranking in the Google search engine, and Kleinberg's HITS algorithm, which analyzes a local neighborhood of the web graph containing an initial set of web pages matching the user's query. Since that time, several other linked-based methods for ranking web pages have been proposed including variants of both PageRank and HITS, and this remains an active research area in which there is still much fertile research ground to be explored. Besides just looking at the link structure in web pages, it is also possible to exploit the anchor text contained in links as an indication of the content of the web page being pointed to. Especially since anchor text tends to be short, it often gives a concise human generated description of the content of a web page. By harnessing anchor text, it is possible to have index terms for a web page even if the page contains only images. Determining which terms from anchors and surrounding text should be used in indexing a page presents other interesting research venues.

2.1 Adversarial Classification: Dealing with Spam on the Web

One particularly intriguing problem in web IR arises from the attempt by some commercial interests to unduly heighten the ranking of their web pages by engaging in various forms of spamming. One common method of spamming involves placing additional keywords in invisible text on a web page so that the page potentially matches many more user queries, even if the page is really irrelevant to these queries. Such methods can be effective against traditional IR ranking schemes that do not make use of link structure, but have more limited utility in the context of global link analysis. Realizing this, spammers now

also utilize link spam where they will create large numbers of web pages that contain links to other pages whose rankings they wish to raise. Identifying such spam in both text-based and linked-based analyses of the web are open problems where AI techniques such as Natural Language Processing (NLP) and Machine Learning (ML) can have a direct impact. For example, statistical NLP methods can be used to determine the likelihood that text on a web page represents "natural" writing. Similarly, classification methods can be applied to the problem of identifying "spam" versus "non-spam" pages, where both textual and non-textual information can be used by the classifier. Especially interesting is that such classification schemes must work in an adversarial context as spammers will continually seek ways of thwarting automatic filters. Adversarial classification is an area in which precious little work has been done, but effective methods can provide large gains both for web search as well as other adversarial text classification tasks such as spam filtering in email.

2.2 Evaluating Search Results

Even when advances are made in the ranking of search results, proper evaluation of these improvements is a non-trivial task. In contrast to traditional IR evaluation methods using manually classified corpora such as the TREC collections, evaluating the efficacy of web search engines remains an open problem and has been the subject of various workshops. Recent efforts in this area have examined interleaving the results of two different ranking schemes and using statistical tests based on the results users clicked on to determine which ranking scheme is "better". There has also been work along the lines of using decision theoretic analysis as a means for determining the "goodness" of a ranking scheme. Commercial search engines often make use of various manual and statistical evaluation criteria in evaluating their ranking functions. Still, principled automated means for large-scale evaluation of ranking results are wanting, and their development would help improve commercial search engines and create better methodologies to evaluate IR research in broader contexts.

3. Using the Web to Create "Kernels" of Meaning

Another challenge in web search is determining the relatedness of fragments of text, even when the fragments may contain few or no terms in common. In our experience, English web queries are on average two to three terms long. Thus, a simple measure of similarity, such as computing the cosine of the terms in both queries, is very coarse and likely to lead to many zero values. For example, consider the fragments "Captain Kirk" and "Star Trek". Clearly, these two fragments are more semantically similar than "Captain Kirk" and "Fried Chicken", but a simple term-based cosine score would give the same (zero) value in both cases. Generalizing this problem, we can define a real-valued kernel function K(x, y), where x and y are arbitrary text fragments. Importantly, we note that K can utilize external resources, such as a search engine in order, to determine a similarity score1.

To this end, we can perform query expansion on both x and y using the results of a search engine and then compute the cosine between these expanded queries. More formally, let QE(t) denote the query expansion of text t, where (for example) we could define QE(t) as the centroid of the TFIDF vector representations of the top 30 documents returned by a search engine in response to query t. We can now define K(x, y) as the cosine between QE(x) and QE(y). Illustratively, we obtain the following results with such a kernel function, anecdotally showing its efficacy:

K("Captain Kirk", "Mister Spock") = 0.49
K("Captain Kirk", "Star Trek") = 0.38
K("Captain Kirk", "Fried Chicken") = 0.02

While such a web contextual kernel function has obvious utility in determining the semantic relatedness of two text fragments by harnessing the vast quantities of text on the web, open research issues remain. For example, future research could help identify more effective text expansion algorithms that are particularly well suited to certain tasks. Also, various methods such as statistical dispersion measures or clustering could be used to identify poor expansions and cases where a text fragment may have an expansion that encompasses multiple meanings.

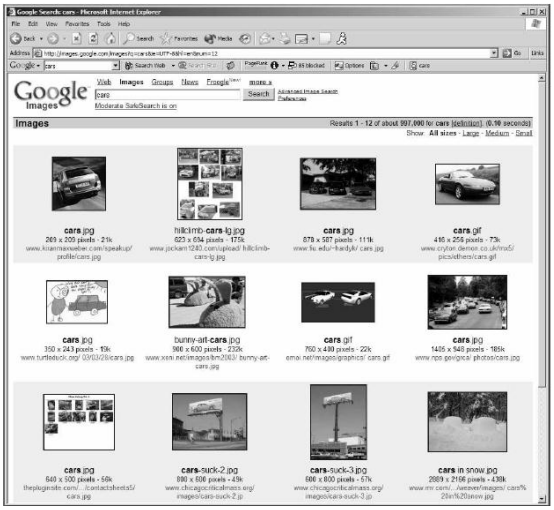
4. Retrieval of UseNet Articles

One of the less visible document collections in the context of general purpose search engines is the UseNet archive, which is conservatively estimated to be at least 800 million documents. The UseNet archive, mostly ignored in traditional academic IR work—with the one exception of the 20 newsgroups data set used in text classification tasks—is extremely interesting. UseNet started as a loosely structured collection of groups that people could post to. Over the years, it evolved into a large hierarchy of over 50,000 groups with topics ranging from sex to theological musings. IR in the context of UseNet articles raises some very interesting issues. As in the case of the Web, spam is a constant problem. However, unlike the web, there is no clear concept of a home page in UseNet. For example, what should the canonical page for queries such as "IBM" or "Digital Cameras" be? One previously explored possibility is to address retrieval in UseNet as a two stage IR problem:
(1) find the most relevant newsgroup, and
(2) find the most relevant document within that newsgroup.

While this may appear to be a simple scheme, consider the fact that there are at least 20 newsgroups that contain the token "IBM". This leads us to the problem of determining whether the canonical newsgroup should be based on having "IBM" at the highest level the group with the most subgroups underneath it, or simply the most trafficked group. Still, other questions arise, such as whether moderated newsgroups should give more weight that un moderated newsgroups or if the Big-8 portion of the UseNet hierarchy should be considered more credible than other portions. At the article or posting level, one can similarly rank not just by content relevance, but also take into account aspects of articles that not normally associated with web pages, such as temporal information, thread information, the author of the article, whether the article quotes another post, whether the proportion of quoted content is much more than the proportion of original content, etc. Moreover, recognizing that certain postings may be FAQs or "flames" would also aid in determining the appropriate ranking for an article. Along these lines, previous research has examined building models of newsgroups, communication patterns within message threads,

and language models that are indicative of content. Still, questions remain of how to go about using such factors to build an effective ranking function and how to display these results effectively to users.

Furthermore, one can also attempt to compute the inherent quality or credibility level of an author independent of the query, much as PageRank does for the Web. Such a computation would operate on a graph of relatively modest size since, for example, if we were to filter authors to only those that had posted at least twice in a year to the same newsgroup, we would be left with only on the order of 100,000 authors. This is a much more manageable size than the web graph which has several billion nodes. Computing community structures—rather than pure linear structures as in posting threads—can also generate interesting insights as to how various authors and groups participate in and influence discussions. One of the most comprehensive studies on bulletin board postings is the Netscan project. This work examined characteristics of authors and posting patterns, such as identifying characteristics of people who start discussions, people who "flame", people who cross-post to multiple newsgroups, people who spam, people who seem to terminate threads, etc. More recently, work on filtering technologies in the context of information retrieval has also focused attention on building better models of the likely content in messages and routing them to appropriate people, bringing together work on user modeling, IR, and text analysis. An advantage of working with the UseNet archive is the fact that it alleviates many of the infrastructural problems that might otherwise slow research in the web domain, such as building HTML parsers, properly handling different languages and character sets, and managing the exceptional volume of available data. Contrastingly, much of the older UseNet posting archive was previously available on a few CD-ROMs, making the archive relatively easy to store, index and process on a single machine. More recently, researchers have started looking at an even smaller scale problem: culling information from bulletin board postings and trying to ascribe a quality level to the information contained therein. For example, Arnt and Zilberstein analyzed postings on the Slashdot bulletin board, attempting to learn the moderation system used. Slashdot moderators assign both a genre label— such as "informative", "funny", etc.—and a score between -1 and +5 indicating their view on how relevant a posting is. Given these score and label pairs, it is a challenging task to use the rich structure of the domain to predict both the label and score for new postings. More generally, improving ranking methods for UseNet or bulletin board postings is an open area of research with many interesting similarities to the web, but also with very many significant differences that make it a fascinating subject of further study.



Results obtained by Searching Google image for "Cars"

5. Retrieval of Images and Sounds

With the proliferation of digital still and video cameras, camera phones, audio recording devices, and mp3 music, there is a rapidly increasing number of non-textual "documents" available to users. One of the challenges faced in the quest to organize and make useful all of the world's information, is the process by which the contents of these non-textual objects should indexed. An equally important line of study (although not a focus of this paper) is how to present the user with intuitive methods by which to query and access this information. The difficulties in addressing the problem of non-textual object retrieval are best illustrated through an example. Figure 1 shows 12 results obtained by searching Google's image repository for "cars". Note the diverse set of content related to cars that is present. In the first 12 results, we see everything from different car poses, pictures of cars on billboards, cars barely visible through the snow, cars for parades, and even hand drawn illustrations. In addressing this sort of diversity, we presently give three basic approaches to the task of retrieving images and music.

1. Content Detection: For images, this method means that the individual objects in the image are detected, possibly segmented, and recognized. The image is then labeled with detected objects. For music, this method may include recognizing the instruments that are played as well as the words that are even determining the artists. Of the three approaches, this is the one that is the furthest from being adequately realized, and involves the most signal processing.

2. Content Similarity Assessment: In this approach, we do not attempt to recognize the content of the images. Instead, we attempt to find images that are similar to the query items. For example, the user may provide an image of what the types of results that they are interested in finding, and based on low-level similarity measures, such as color histograms, audio frequency histograms, etc, similar objects are returned. Systems such as these have often been used to find images of sunsets, blue skies, etc. and have also been applied to the task of finding similar music genres.

3. Using Surrounding Textual Information: A common method of assigning labels to non-textual objects is to use

information that surrounds these objects in the documents that they are found. For example, when images are found in web documents, there is a wealth of information that can be used as evidence of the image contents. For example, the site on which the image appears, how the image is referred to, the image's filename, and even the surrounding text all provide potentially relevant information about the image.

All of these approaches can, of course, be used in conjunction with each other, and each provides a fairly diverse set of benefits and drawbacks. For example, surrounding textual information is the easiest method to use; however it is the most susceptible to misclassification of the image content, due to both errors and malicious web site designers. Content Similarity Assessment can provide some indication of the image content, but is rarely able in practice to find particular objects or particular people. Content Detection is the only method that attempts to recognize the objects in the scene; however, building detectors for arbitrary objects is a time consuming task that usually involves quite a bit of custom research for each object. For example, the most studied object detection domain to date is finding faces in images, and work has continued on improving the quality for almost a decade. Work in using these systems to detect people and cars are progressing; extending to arbitrary objects is also the focus of a significant amount of research. Finally, looking into the future, how many of these ideas can be extended to video retrieval? Combining the audio track from videos with the images that are being displayed may not only provide additional sources of information on how to index the video, but also provide a tremendous amount of (noisy) training data for training object recognition algorithms en masse.

6. Harnessing Vast Quantities of Data

Even with the variety of research topics discussed previously, we are only still scratching the surface of the myriad of issues that AI technologies can address with respect to web search. One of the most interesting aspects of working with web data is the insight and appreciation that one can get for large data sets. This has been exemplified by Banko and Brill in the case of word sense disambiguation, but as a practical example, we also briefly discuss our own experiences in two different contexts at Google: Spelling Correction and Query Classification.

Spelling Correction. In contrast to traditional approaches which solely make use of standard term lexicons to make spelling corrections, the Google spelling corrector takes a Machine Learning approach that leverages an enormous volume of text to build a very fine grained probabilistic context sensitive model for spelling correction. This allows the system to recognize far more terms than a standard spelling correction system, especially proper names which commonly appear in web queries but not in standard lexicons. For example, many standard spelling systems would suggest the text "Mehran Sahami" be corrected to "Tehran Salami", being completely ignorant of the proper name and simply suggesting common terms with small edit distance to the original text. Contrastingly, the Google spelling corrector does not attempt to correct the text "Mehran Sahami" since this term combination is recognized by its highly granular model. More interesting, however, is the fact that by employing a context sensitive model, the system will correct the text "Mehran Salhami" to "Mehran Sahami" even though "Salami" is a common English word and is the same edit distance from "Salhami" as "Sahami." Such fine grained context sensitivity can only be achieved through analyzing very large quantities of text.

Query Classification into the Open Directory Project. The Open Directory Project (ODP) is a large open source topic hierarchy into which web pages have been manually classified. The hierarchy contains roughly 500,000 classes/topics. Since this is a useful source of hand-classified information, we sought to build a query classifier that would identify and suggest categories in the ODP that would be relevant to a user query. At first blush, this would appear to be a standard text classification task. It becomes more challenging when we consider that the "documents" to be classified are user queries, which have an average length of just over two words. Moreover, the set of classes from the ODP is much larger than any previously studied classification task, and the classes are non-mutually exclusive which can create additional confusion between topics. Despite these challenges, we have available roughly four million pre-classified documents, giving us quite a substantial training set. We tried a variety of different approaches that explored many different aspects of the classifier model space: independence assumptions between words, modeling word order and dependencies for two and three word queries, generative and discriminative models, boosting, and others. The complete list of methods compared is not included since some portions of the study were conducted in an iterative piecemeal fashion, so a direct comparison of all methods applied to all the data is not possible to provide. Nevertheless, we found that the various algorithms performed as expected relative to previously published results in text classification when training data set sizes were small. Interestingly, as we steadily grew the amount of data available for training, however, we reached a critical point at which most of the algorithms were generally indistinguishable in performance. Furthermore, most probability smoothing techniques, which generally seem to help in limited data situations, either showed no appreciably improvements or actually decreased performance in the data rich case for Naïve Bayes. While the set of alternative algorithms used was by no means exhaustive, and the results here are still somewhat anecdotal, we hypothesize that, as in the case of the Banko and Brill study, an abundance of data often can, and usually does, make up for weaker modeling techniques. This perspective can be unusually liberating—it implies that given enough training data, the simpler, more obvious solutions can work, perhaps even better than more complex models that attempt to compensate for lack of sufficient data points.

7. Conclusions

Web information retrieval presents a wonderfully rich and varied set of problems where AI techniques can make critical advances. In this paper, we have presented a number of challenges, giving an overview of some approaches taken toward these problems and outlining many directions for future work. As a result, we hope to stimulate still more research in this area that will make use of the vast amount of information on the web in order to better achieve the goal of organizing the world's information and making it universally accessible and useful.

References:

[1] Wu, J., Rehg, J.M., Mullin, M.D.: Learning a Rare Event Detection Cascade by Direct Feature Selection. In: Advances in Neural Information Processing Systems 16, 2004
[2] Berenzweig, A., Logan, B., Ellis, D., Whitman, B.: A Large-Scale Evaluation of Acoustic and Subjective Music Similarity Measures. In: Proc. of the 4th International Symposium on Music Information Retrieval, 2003
[3] Dumais, S., Bharat, K., Joachims, T., Weigend, A. (eds.): Workshop on Implicit Measures of User Interests and Preferences at SIGIR, 2003
[4] Arnt, A., and Zilberstein, S.: Learning to Perform Moderation in Online Forums. In: Proc. of the IEEE/WIC International Conference onWeb Intelligence, 2003
[5] Viola, P., Jones, M., Snow, D.: Detecting Pedestrians Using Patterns of Motion and Appearance. Mitsubishi Electric Research Lab Technical Report. TR-2003-90, 2003
[6] Tomlin, J.A.: A New Paradigm for Ranking Pages on the World Wide Web. In: Proc. of the 12th International World Wide Web Conference, 350-355, 2003
[7] Zhang, Y., Callan, J., Minka, T.P.: Novelty and Redundancy Detection in Adaptive Filtering. In: Proc. of the 25th International ACM-SIGIR Conference on Research and Development in Information Retrieval, 2002
[8] Fiore, A., Tiernan, S.L., Smith, M.: Observed Behaviour and Perceived Value of Authors in Usenet Newsgroups: Bridging the Gap, In: Proc. of the ACM SIGCHI Conference on Human Factors in Computing Systems, 323-330, 2002
[9 Agosti, M., and Melucci, M. (eds.): Workshop on Evaluation of Web Document Retrieval at SIGIR, 1999

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Improve the Classification and Prediction Performance for the IP Management System in a Super-capacitor Pilot Plant

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***Abstract*:** This paper examines the performance of support vector machines in the Super-capacitor Pilot Plant IP Management System. In our system model, two functions made by the inference engine (support vector machines)-classifying information which is not only obtained from external sources such as the web, but also obtained directly from a working super-capacitor pilot plant (internal source), predicting cases which has been organized in the knowledge base by the classification functions. Our tests analyze four common inner-product kernels of the support vector machine through two groups of sample dataset (small size 0-1000 attributes and large size 1000-4000 attributes). In particular, we compare four conventional classifiers for which are commonly used as retrieval engine in the case based reasoning cycle. Overall, we find strong evidence of the classification and prediction ability of support vector machines in the proposed IP management system. Our tests support the two hypotheses that using support vector machines in the proposed intelligentized intellectual property system is likely lead to a better classified and predictive performance.

***Keywords*:** Performance Study, Intelligent Intellectual Property Management System, Data Mining, Case-Based Reasoning and Support Vector Machine.

1. Introduction

From the technological point of view, the aim of our intellectual property management system is looking for ways to facilitate the efficient and complete retrieval of relevant information from internal (i.e., to a company or establishment) and external sources (i.e., web or external database) with a high degree of accuracy; to set up inner IP Model component concerning internal intellectual property rights and IP knowledge base component with regard to global intellectual property information; to design and implement the artificial intelligence methodology in order to facilitate the establishing, retrieving and comparing these two components. In our system model, two functions made by the inference engine (support vector machines)-classifying information which is not only obtained from external sources such as the web, but also obtained directly from a working super-capacitor pilot plant (internal source), predicting cases which has been organized in the knowledge base by the classification functions. Therefore we need to design experiments to compare common inner-product kernels of the support vector machine with conventional classifiers through different sample size. The aim is to find the evidence that our proposed engine is capable of improving the classification and prediction performance of the proposed IP management system.

Basically, based on the data mining concept [1] [2] machine learning theory [3] and case-based reasoning algorithm [4], we proposed an intelligentized intellectual property management system model which implementing a hybrid data mining and case-based reasoning architecture to help with identifying, managing, protecting and exploiting valuable intellectual property automatically and intelligently in real time for the supercapacitor pilot plant. The system architecture is shown in Figure 1. This figure outlines the general framework from the perspective of its system architecture. The hybrid system was integrated with four levels: a) User Level b) Decision Making Level c) Data Mining Level d) Resource Level to achieve the IP intelligentized management functions.

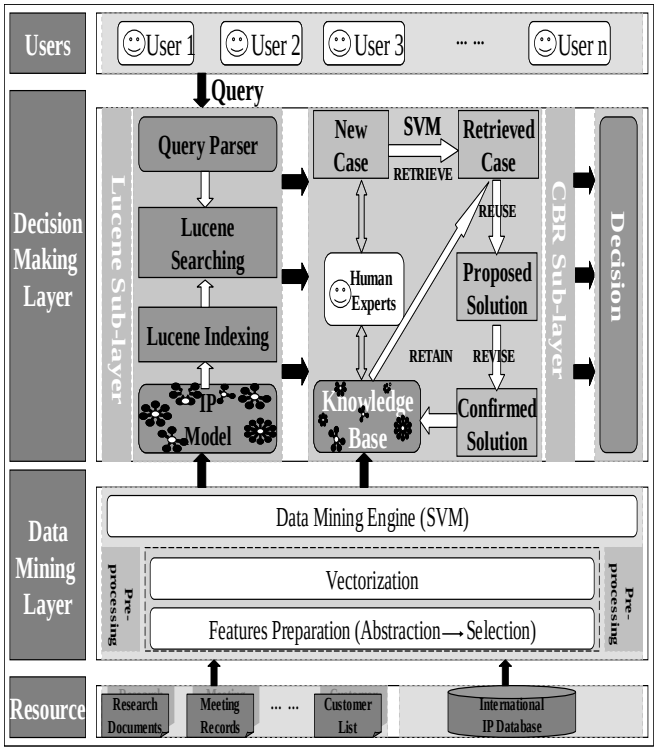

Figure 1. System architecture

2. Theoretical Framework and Hypothesis

2.1 Data Mining Layer

In 1989 G. P. Shapiro [1] defined the standard knowledge discovery in database (KDD) process to prove data mining was central to successfully extract knowledge in data. Building on this, we hypothesized that data mining process will ensure the establishment of inner IP model and IP knowledge base for our system.

The original KDD process was modified to fit our supercapacitor pilot plant by:

- Replacing target data preprocessed data and transformed data with feature and vector, because they are more accurate and suitable for the data mining engine;
- Adding support vector machine to serve as data mining engine.

The high-level goals of data mining layer tend to be descriptive, predictive, or a combination of predictive and descriptive [5]. In our hybrid system, this layer is to achieve the descriptive goal, focuses on understanding the underlying data; while the case-based reasoning sub-layer is for the predictive goal focuses on testing the accuracy in the descriptive ability which is going to be introduced in the next section.

2.2 Case-Based Reasoning Sub-layer

The basic premise of machine learning is that data preserving previous decision experience exist [3]. These experience-based patterns are called cases which have been prepared by the data mining layer. Case-based reasoning uses these patterns to support similar decision. The foundation of this algorithm is the knowledge base containing a number of pervious cases for decision-making. Researches [6] have now proved case-based reasoning approach is extremely effective for problems in which the rules are inadequate same as our pilot plant.

The original case-based reasoning cycle has been modified by:

- Loading the retrieve reuse revise retain functions into the task pool;
- Adding method library;
- Using SVM to serve as retrieval engine;
- Treating user's query as a new case, computing similarities between query and cases.

A retrieval engine [7] should retrieve the most similar cases to the current query (new case) which relies on directly searching the knowledge base to get the potentially useful cases. Only when retrieval engine is efficient at handling thousands of cases the Case-based reasoning will be applicable for large scale problems. Comparing with the database searching that specific value in a record, retrieval of cases from the case-base must be equipped with heuristics that perform partial matches, because in general there is no existing case that exactly matches the new case [7].

Well known methods for case retrieval in the case-based reasoning cycle are [8, 9] from artificial intelligence (AI), information retrieval (IR), database (DB), and statistics (STAT). These typical classifiers include the K-Nearest Neighbors (K-NN), the Naïve Bayes (NB), the Random Decision Tree (RDT) and the One Rule-based reasoning (ORBR) [10].

In our method, support vector machines (SVMs) [11] are implemented here again to serve as retrieval engine in this sub-layer; theoretically, the training session of SVMs corresponds to the data mining process which is used for creating IP models and knowledge base, while the retrieval function of the case-based reasoning can be looked as the testing session of SVMs which is used for testing of the usability of IP models and knowledge base.

2.3 Support Vector Machines

The Support Vector Machine is a universal feed-forward network, pioneered by Vapnik [11]. The promising application for this technique is data classification and retrieval.

From the learning theory perspective, the support vector machine has the inherent ability to solve the pattern classification problem in a manner close to the optimum for the problem of interest [11]. Moreover, it is able to achieve a remarkable performance with no problem domain knowledge built into the design of the machine which is more applicable for the intellectual property management problem in our supercapacitor pilot plant case.

The support vector machine is built as follows:

In accordance with Cover's theorem [12] which is a complex pattern classification problem cast in a high dimensional space nonlinearly is more likely to be linearly separable than in a low dimensional space;

- Nonlinear mapping of an input vector into a high dimensional attributes space that is hidden from both the input and output.

Following the structural risk minimization principle [12], that is the error rate of a learning machine on test data is bounded by the sum of the training-error rate and VC dimension [12];

- Construction of an optimal maximum margin hyper plane for separating attributes discovered in step1 as the decision surface, producing a value of zero for the first term and minimizing the second term.

Accordingly, the VC dimension is minimized and good generalization performance on data classification problems which arose in the establishment process for the IP model and global IP knowledge base is achieved. In term of learning process, this is the training session, during which the support vector machine is repeatedly presented a set of input vector along with the category to which each particular pattern belongs.

Therefore, we expected that,

Hypothesis 1 Using support vector machine to serve as data mining engine in the proposed intelligentized intellectual property management system is likely lead to a better classified performance.

On the other hand, in the case-based reasoning cycle, a new case is presented to the support vector machine that has not seen before, but which might belong to the same population of patterns used to train the support vector machine. The support vector machine is able to identify the class of that new case because of the information it has extracted from the training

session.

Therefore, we expected that,

Hypothesis 2 Using support vector machine to serve as case-based reasoning retrieval engine in the proposed intelligentized intellectual property management system is likely lead to a better predictive performance.

3. Method

3.1 Experiment Design

Since we have hypothesized the support vector machine which served as data mining engine and case-based reasoning retrieval engine will likely get the better performance, to test it two different series of experiments are considered:

- Do the weighting of four common inner-product kernels of the support vector machine [11]: Poly Kernel, Normalized Poly Kernel, Puk and Radial Basis Function (RBF) Kernel; each with two groups of sample dataset, small size sample and large size sample.
- Do the weighting of four conventional classifiers [13] the K-Nearest Neighbors (K-NN), the Naïve Bayes (NB), the Decision Tree (DT) and the Rule based reasoning function, each with the same two groups of sample dataset, small size sample and large size sample.

The tool used in these experiments is Weka [14], which is a machine learning algorithms workbench. The overall performance measures [15] [16] used in our study are: root means, squared error, root relative squared error, mean absolute error, relative absolute error, and kappa statistic. Detailed performance measure: the true positives (TP), the false positive (FP), precision, recall, F-measure and receiver operating characteristic (ROC) are choose to test the specified class (C1).

According to the Sample Size Determination Table in [17], two groups of sample have been selected in our experiments, small size sample have attached attribute ranging from 0001 to 1000, and we choose a random start data point at 200 attribute, and then pick every 200th attribute, thereafter give us our sample 400 attribute, 600 attribute, 800 attribute; large size sample have attached attribute ranging from 1000 to 4000, and we chose a random start data point at 1000 attribute and then pick every 1000th attribute, thereafter took 2000 attribute, 3000 attribute, 4000 attribute.

Table 1. Experiment results with 200 attribute

Classifiers	200 Attribute							
	Poly	**NPoly**	**Puk**	**RBF**	**KNN**	**NB**	**RDT**	**ORBR**
Correctly Classified	0.9565	0.9275	0.7246	0.6572	0.8116	0.5652	0.5072	0.2464
Seconds to build Model	3.3300	3.4100	6.4200	3.3000	0.0500	0.000	0.1700	0.0500
Root Mean Squared Error	0.3012	0.3027	0.3239	0.3250	0.3491	0.3525	0.3752	0.4640
Root Relative Squared Error	0.8606	0.8649	0.9254	0.9287	0.9325	1.0072	1.0720	1.3259
Mean Absolute Error	0.2047	0.2051	0.2183	0.2208	0.0729	0.1246	0.1408	0.2153
Relative Absolute Error	0.8357	0.8374	0.8913	0.9018	0.2976	0.5086	0.5749	0.8792
Kappa Statistic	0.9493	0.9154	0.6780	0.5934	0.7800	0.4924	0.4250	0.1186
Class	**C3**							
TP	0.9000	0.9000	0.7000	0.5000	0.8000	0.6000	0.5000	0.4000
FP	0.0170	0.0340	0.0170	0.0000	0.0340	0.1020	0.1020	0.3220
Precision	0.9000	0.8180	0.8750	1.000	0.8000	0.5000	0.4550	0.1740
Recall	0.9000	0.9000	0.7000	0.5000	0.8000	0.6000	0.5000	0.4000
F-measure	0.9000	0.8570	0.7780	0.6670	0.8000	0.5450	0.4760	0.2420
Roc Area	0.9900	0.9780	0.9790	0.9350	0.8980	0.8100	0.6990	0.5390

3.2 Experiment Ⅰ

In this series of experiments, we compared the following four kernel types [11] of the support vector machine:

- Poly Kernel
- Normalized Poly Kernel
- Puk
- Radial Basis Function (RBF) Kernel

For each of these kernels we implemented the learning process. Table 1 presents the experiment result for small size sample at the random start data point (200 attribute), two groups of measure were applied, one was the overall performance measurement of the support vector machine and another was the detailed accuracy performance measurement by a specified class (i.e., C3). The experiment results for large size sample at the random start data point (1000 attribute) with the same measurements in Table 1 was shown in Table 2. As for other data point 400 attribute 600 attribute 800 attribute 2000 attribute 3000 attribute 4000 attribute, the accuracy experiment result can be found in Figure 2.

WEKA provides several options for testing the results of the IP model and knowledge base. In our study we tested them on the training data by using cross-validation [18] method and indicated folds to 10, which means the IP model will be tested ten times by:

- Holding out 1/10 of the training data set
- Developing a model for the remaining 9/10 of the training data set
- Testing the resulting IP model on the 1/10 withheld.

The data withheld is selected at random from the data not yet tested, at the conclusion all data will have been used as test data, so that the testing accuracy in the experiment have been assured.

As far as other parameters, all followed the recommendation from the Weka too. For all these kernels the cache size was set to 250007, the number of kernel evaluations was 210, support vectors was 20. The parameter of the support vector machine was set to 1.0. In table 2, the exponent was denoted to 2.0 for the Normalized Poly Kernel, and it was 1.0 for the Poly kernel. In the Puk kernel and was set to 1.0. In the Radial Basis Function (RBF) Kernel, was set to 0.01.

This study attempts to understand the competence of different kernel function within the support vector machine in determining the performance of classification and prediction.

Table 2. Experiment Results with 1000 attribute

Classifiers	1000 Attribute							
	Poly	**NPoly**	**Puk**	**RBF**	**KNN**	**NB**	**RDT**	**ORBR**
Correctly Classified	0.8406	0.6812	0.6232	0.5942	0.4783	0.3623	0.2899	0.2464
Seconds to build Model	3.4200	2.8800	1.7700	1.7800	0.0000	0.2000	0.7700	0.3000
Root Mean Squared Error	0.3065	0.3169	0.3317	0.3262	0.3677	0.4268	0.4504	0.4640
Root Relative Squared Error	0.8758	0.9056	0.9477	0.9322	1.0508	1.2197	1.2871	1.3259
Mean Absolute Error	0.2074	0.2139	0.2236	0.2212	0.1588	0.1822	0.2029	0.2153
Relative Absolute Error	0.8470	0.8376	0.9130	0.9034	0.6484	0.7440	0.8285	0.8792
Kappa statistic	0.8140	0.6276	0.5594	0.5255	0.3907	0.2557	0.1707	0.1186
Class	**C3**							
TP	0.7000	0.6000	0.4000	0.6000	0.6000	0.7000	0.6000	0.5000
FP	0.0340	0.0170	0.0170	0.0170	0.0170	0.1860	0.0850	0.4580
Precision	0.7780	0.8570	0.8000	0.8570	0.8570	0.3890	0.5450	0.1560
Recall	0.7000	0.6000	0.4000	0.6000	0.6000	0.7000	0.6000	0.5000
F-measure	0.7370	0.7060	0.5330	0.7060	0.7060	0.5000	0.5710	0.2380
Roc Area	0.9780	0.8720	0.7800	0.8810	0.7590	0.7320	0.7580	0.5210

3.3 Experiment Ⅱ

In this series of experiments, Conventional classification methods [13] in the case-based reasoning cycle tested in our research included:

- K-Nearest Neighbors (K-NN)
- Naïve Bayes (NB)
- Random Decision Tree (RDT)
- One Rule Base Reasoning (ORBR)

We evaluated the different methods on the experimental dataset which was same in Experiment I. Table 1 and 2 showed the evaluation experiment result concerning the overall performance of the four conventional classifiers and also the

detailed predictive accuracy performance by a specified class (i.e., C3) according to the random start data point in different size of sample.

During the test process, cross-validation method was used and folds were indicated to 10 same as it was in experiment Ⅰ. Other parameters all followed the basic setting provided by the Weka.

For the K-Nearest Neighbors, the number of neighbors to use was set to 1. In the Random Decision Tree, the number of randomly chosen attribute (KValue) was denoted to 1; the maximum depth of the tree was unlimited; the minimum total weight of the samples in a leaf was set to 1.0; the random number seed used for selecting attribute was denoted to 1. As for the One Rule Based Reasoning method, the minimum bucket size for discretizing the attribute was chosen to 6.

This study attempts to understand the ability of the typically used retrieval engine within the case-based reasoning cycle for determining the performance of classification and prediction.

4. Results and Conclusions

In Figure 2, we showed the overall performance, as measured by classified-accuracy, of all methods at 8 data point in two groups of different attribute sample size. We tuned each method (support vector machines with four different kernels and convention classifier) to the basic set recommended by the Weka.

The averaged experiments results over all 200 and 1000 attributes in Table 1 and 2 and Figure 2 show two major findings:

(1) In the whole class classification task, Poly kernel is the best learning function for both small and large size sample classification task. Support vector machines with all kernels almost outperform all conventional classifiers, except K-Nearest Neighbors in 200 attribute data point which is better than Puk and RBF kernels but worse than Poly and NPoly kernels.

(2) In the specified class (C3) prediction task, RBF kernel is the best learning function for both small and large attribute sample size. Although K-Nearest Neighbors also provide best result for the same task in the large size sample, considering other measurements (Kappa Statistic and Roc Area), support vector machines with all kernels outperform conventional classifiers

These empirical results provided evidence for two Hypotheses. First, let us consider the overall performance. In Figure 2, we grouped all these methods by accuracy in order to find out whether there was difference in performance between the support vector machine with different kernels and conventional classifiers. A good classification and prediction method, we would expect, has about equal better performance for different attribute sample size by either side. Comparing accuracy performance for all these methods in our experiments, shown in Fig. 7, we found that only the support vector machine with different kernels satisfied this requirement. Poly kernel achieved the best performance with 95.65% accuracy at 200 attributes data point and 56.52% at 4000 attribute data point; followed by NPoly kernel, with 92.75% at 200 attributes data point and 53.62% at 4000 attributes data point, even the RBF kernel which got the worst performance in the kernels group reached 43.48% at 4000 attribute data point. While for the best conventional classifiers K-Nearest Neighbors, although obtained 81.16% accuracy at 200 attribute data point, when the attribute sample size increased to 4000, its performance accuracy dropped dramatically to 30.43%

In conclusion, support vector machines with all kernels outperform the best conventional classifiers K-Nearest Neighbors (1 time better in total 32 times comparison) by up to 96.88%. In Table 4 and 5, focusing on the root mean squared error and root relative squared error, the Poly kernel was also the best function which had lowest value of these two error rate measure, that is 0.3012 and 0.8606 at 200 attributes data point; 0.3065 and 0.8758 at 1000. While the value of the best conventional classifiers K-Nearest Neighbors was 0.3491 and 0.9325 at 200; and 0.3677 and 1.0508 at 1000. Since error rate measurement is negatively-oriented scores: Lower values are better, Poly kernel was the best function whether in support vector machines group or in conventional classifiers group. As for mean absolute error and relative absolute error, things were slightly different, in experiment Ⅰ indicated that the differences among all the four kernels were statistically significant. For the results with Poly kernel, lowest for both of these two measurements whether at 200 attributes data point or 1000 attributes data point. And in experiment Ⅱ, for these two measures the K-Nearest Neighbor got the lowest value and the differences among all the conventional classifiers were statically significant too. One question is that these two error rate measurements in the group of support vector machines with different kernels were much higher than most of the conventional classifiers, such as K-Nearest Neighbor, the Naive Bayes and the Random Decision Tree, which is not consistent with two Hypotheses. Actually, these two differences are not statistically significant when comparing between machine learning methods and traditional classification methods. The observed experiments results of correctly classified measurement in Table 4 and 5 provided the evidence that such two kinds of measurement are not ideal for our task.

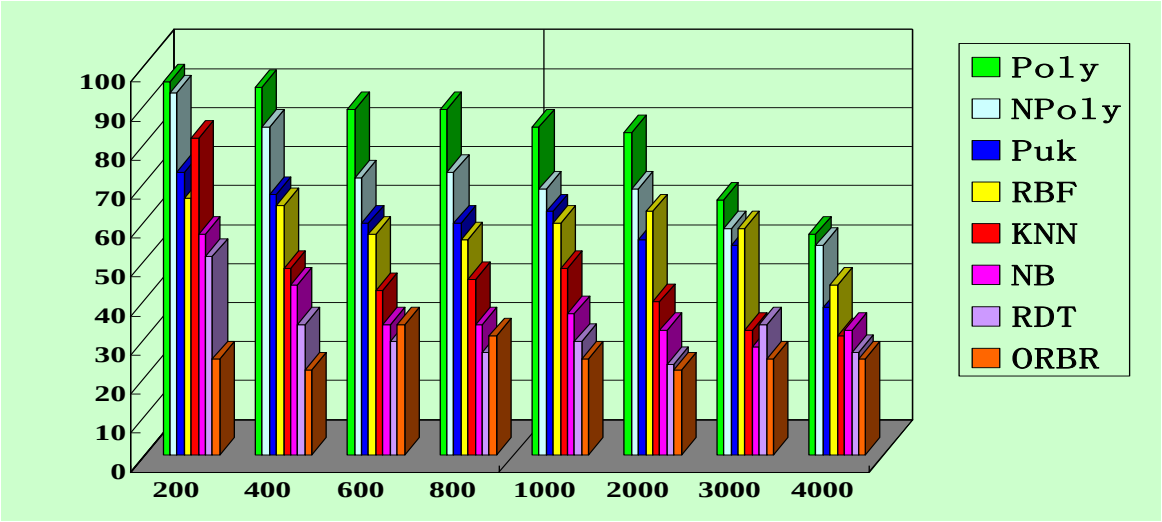

Figure 2. Accuracy comparison result

Secondly, in order to see if the accuracy of different methods for a selected class (C3) is still reliable, detailed accuracy experiments were performed. Using TP FP Precision and Recall to calculate F-measure and Roc Area, as expected, support vector machines with different kernels was observed a superior accuracy performance whether at 200 or 1000 attribute data point. Experiments result in Table 4 and 5; showed for a specified class prediction task, support vector machines with all kernels outperformed the best conventional classifiers K-Nearest Neighbors (1 time better in total 8 times comparison) by up to 75 %. The interesting phenomenon was the RBF kernel, the worst overall kernel performance in the support vector machine group, had the greatest precision for predicting the given class C3 for both attribute sample size. In particular at 200 attributes data point it reached 100% precision. This saturation was interpreted as additional evidence for the Hypothesis II. Furthermore, as expected, Poly kernel achieved the best F-measure and Roc Area that was 0.9000 and 0.9900 at 200 attributes data point; 0.7370 and 0.9780 at 1000 attribute data point. For the K-Nearest Neighbors, got 0.8000 and 0.8980 at 200 attribute data point, but after the attribute increased to 1000, we saw a drop to 0.706 and 0.7590. From all these results, it appeared clearly that nearly all of these kernel approaches achieved the larger F-measure and Roc Area. More surprisingly was the absolute superiority in the Roc Area of all kernels at both attribute sample size. Roughly speaking, the larger the ROC area is, the better the classified performance. Additionally, Kappa Statistic, which is a statistical measure of overall inter-class items, suggests kernel functions are more reliable than conventional classifiers. Poly kernel got the Kappa Statistic value of 0.9493 at 200 attribute data point and 0.8140 at 1000 attribute data point; the best conventional classifiers K-Nearest Neighbors got 0.7800 at 200 attribute data, but when the attribute sample size reached 1000, the value fell dramatically to 0.3907.

Overall, the experiment results were encouraging in that supporting the two hypotheses. The support vector machine was found to assist in data mining layer and case-based reasoning sub-layer effectively and reliantly which was capable of classifying and predicting relevant case for new input IP case.

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References

[1] U. M. Fayyad, G. Piatetsky-Shapiro and P. Smyth, “From data mining to knowledge discovery: an overview. Advances in knowledge discovery and data mining”, American Association for Artificial Intelligence, Menlo Park, CA, 1996.

[2] J. Han, M. Kamber, “Data Mining: Concepts and Techniques”, Morgan Kaufmann, academic press, 2001.

[3] T. Mitchell, “Machine Learning”, McGraw-Hill, Boston, MA, 1997.

[4] A. Aamodt, E. Plaza, “Case-based reasoning: foundational issues, Methodological variations, and system approaches”, AI communications, 7(1), pp. 39-59, 1994.

[5] U.Fayyad, G. Piatetsky-Shapiro and P. Smyth, “The KDD process for extracting useful knowledge from volumes of data”, Communications of the ACM, Vol. 39(11), pp.27-34, Nov. 1996.

[6] J. Zeleznikow, “Book review: CASE-BASED REASONING by Janet Kolodner (Morgan Kaufmann Publishers, 1993)”, ACM SIGART Bulletin, Vol. 7(3), pp. 20 - 22, 1996.

[7] I. Watson, F. Marir, “Case-Based Reasoning: A Review. The Knowledge Engineering Review”, Vol. 9(4), 1994.

[8] H. Li, “Case-based reasoning for intelligent support of construction negotiation”, Information & Management, Vol.30, pp. 231-238, 1996.

[9] L. A. Breslow, D. W. Aha, “Simplifying decision trees: A survey”, The Knowledge Engineering Review, Vol.12 Cambridge University Press, 1997.

[10] R.C. Holte, "Very simple classification rules perform well on most commonly used datasets", Machine Learning, Vol.11, pp. 63-91, 1993.
[11] C. Cortes, V. Vapnik. "Support-vector network", Machine Learning, pp.273-297, 1995.
[12] C. M. Bishop, "Neural Networks for Pattern Recognition", Oxford University Press, Inc., New York, NY, 1995.
[13] A. Singhal, "Modern Information Retrieval: A Brief Overview", Bulletin of the IEEE Computer Society Technical Committee on Data Engineering, Vol. 24 (4), pp.35–43, 2004.
[14] I. H. Witten, E. Frank, "Data Mining: Practical machine learning tools and techniques", 2nd Edition, Kaufmann, Morgan, San Francisco, 2005.
[15] E. R. House, "Assumptions underlying evaluation models", Assessment & Evaluation in Higher Education, 1469-297X, Vol. 21(4), pp. 347-356, 1996.
[16] M. Y. Kiang, "A comparative assessment of classification methods", Decision Support Systems, Vol. 35(4), pp. 441 – 454, 2003.
[17] J. E. Bartlett, J. W. Kotrlik and C. Higgins, "Organizational research: Determining appropriate sample size for survey research. Information Technology", Learning, and Performance Journal, Vol. 19(1), pp. 43-50, 2001.
[18] B. Efron, R. Tibshirani, "Improvements on cross-validation: The .632 + Bootstrap Method", Journal of the American Statistical Association, 92 (438), pp. 548-560, 1997.

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Watermarking of h.264 Coded Video Based on the Shifted-Histogram Technique

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***Abstract*:** Reversible video watermarking through shifted histogram of the quantized coefficients of the H.264/AVC coded video is introduced. In CIF sequences, the embedded data of a capacity of more than 2500 bits in I frame is possible and can exceed 7000 bits if we also exploit about 10 P frames in the GOP for the embedding. While the degradation introduced by the watermarked can achieve around 13 dB, the subjective impacts are almost negligible. This data hiding is reversible and it increases the encoded bits by less than 1 % and less than the embedding capacity offering the opportunity to carry information at a lower bitrate rather than a second channel..

***Keywords*:** H264/AVC standard; shifted- histogram technique; video quality assessment; video watermarking.

1. Introduction

The video is often being manipulated in a compressed format for storage and transmission. For various applications that require watermarking techniques such as copyright protection or authentication control, researchers are nowadays moving to the video compressed field and especially towards the latest video codec H264/AVC.

The H264/AVC is today widely adopted and related products and implementations are increasingly appearing. It has demonstrated a high coding efficiency comparing to the previous standards and it becomes then essential to consider the security aspects of this codec. An overview of this standard can be found in reference [1].

Several watermarking techniques were proposed for the H264/AVC video standard, where the embedding is usually performed in I frames, since P and B are highly compressed, and the quantized DCT coefficients are chosen for the insertion because the quantization is a lossy operation, and the signature could be lost if embedded before this step. Though, in most of the proposed methods, even if the video quality and the bitrate are maintained, the embedding capacity is often limited and this can be a real obstacle for some applications.

Indeed, in embedding watermark into a compressed video various watermark requirements such as the embedding capacity and the video quality should be taken into consideration, and for this video standard, increasing the embedding capacity constitute a real challenge rarely addressed in literature. To reach the compromise between video quality and data embedding capacity, the reversible video watermarking schemes seem to be a logical solution for a higher capacity embedding and for recovering the encoded video at the detection step. In this paper we apply a reversible watermarking method based on the shifted-histogram technique to the luma quantized DCT coefficients of the H264/AVC video standard. We evaluate the embedding capacity and the increase in bitrate. The video quality is also assessed before applying any video restoration. Results show that despite heavy degradation on objective video quality of watermarked images, the subjective quality is almost not disturbed by this method.

This paper is organized as follows: in section 2 a presentation of the related work is exposed, in section 3 we describe the application of the shifted histogram technique to the H264/AVC codec and we explain the video quality assessment that has been performed. Results and analysis are exposed in section 4. Three points are essentially discussed: The embedding capacity, the video quality and the increase in bitrate. Finally, we draw conclusions in section 5.

2. Related work

The use of DCT coefficient for embedding the watermark is very common in both compressed image (such as JPEG) and video, this is because compression does not leave much choice as a watermark location. For the H264/AVC video codec, the quantization DCT coefficients and the motion vectors are the most used to insert the signature. Coefficients of low, middle or high frequencies may be selected to embed the watermark and respond to the needs of the application in terms of robustness or fragility. For the embedding process, the choice of the DCT blocks and the DCT coefficients usually relies on the human visual properties to mask the degradation introduced by the watermark. Among these methods, Chen et al. [2] present a video watermarking method based on blocks energy. Two algorithms were proposed for low and high energy blocks of Intra frame, the objective is to consider the high frequency noise attack and the low-pass filter attack. The security of the watermark is ensured by an encryption using a Torus Automorphisms algorithm and the Secret Image Sharing technology to increase robustness. In another example, Noorkami et al. [3] proposed a low comlplexity watermarking method where the watermark is embedded in the AC coefficients of I frames. In order to preserve the video quality, only one coefficient per macroblock is selected to embed the watermark. The security of the algorithm is based on the randomness of the watermark location which is determined by a public key extracted from the feature of the macroblock such as the DC coefficient. A general scheme of the watermarking process based on DCT coefficients is represented in figure 1.

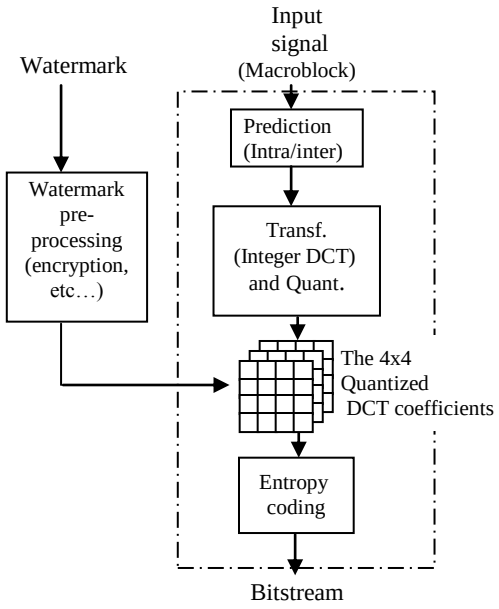


Figure 1. General scheme of watermarking of H264/AVC based on DCT coefficients

Our method is based on the shifted histogram data hiding technique applied to the quantized DCT coefficients. It was initially proposed by Ni et al. [4] for still images and improved later by other authors [5]. The method consists of selecting the peak/ zero pairs from the image histogram and apply a shift between pixels' levels (p,z) of the maximum and the minimum frequencies, so that a gap is created in p. The aim is to maximize the embedding capacity by inserting the watermark in the largest frequency.

The image is scanned after applying the shift, and the watermark bits are hidden as follows:
For L as the pixel level and w as a watermark bit,

1st case: p > z and L=(p-1)
If (w= 1) then L=L+1=p, otherwise,
L= (p-1) (not modified)
2nd case: p < z and L =(p+1)
If (w= 1) then L=L - 1=p, otherwise,
L=(p+1)(not modified)

The data hiding is reversible and detection is done by rescanning the image. If the value of L is equal to p the watermark bit is "1", and if L is equal to p-1 (for the first case) or p+1 (for the second case) the watermark bit is "0".

Several ranges of pairs of peak/zero values could be chosen from the histogram in order to increase the embedding capacity. The values (p,z) need to be identified as side information for a reversible data hiding and need to be transmitted to the receiving side. If the minimum frequency is not equal to zero, the related pixel positions need also to be transmitted as side information [5].

3. Description of the method

3.1 Watermark Insertion/extraction

We have applied the shifted histogram technique to the quantized luma DCT coefficients of I and P frames of the H264 /AVC video sequences. The histogram of the significant DCT coefficients shows almost symmetric positive and negative sides. The following steps describe how to apply the method to the positive side of the histogram:

- Find a pair of maxima and minima (p,z) in the positive significant levels and shift the values within the range [p, z-1] to the right (toward the minimum frequency) by incrementing the values by 1.
- If the Watermark bit is equal to "1", reduce (p+1) by 1 otherwise the coefficient is not modified.

To apply the method in both positive and negative sides, the previous steps are repeated in the negative side of the histogram by shifting the levels within the range [z'+1, p'] to the left. (p',z') is the pair of maxima and minima values in the negative side of the histogram. In our case we have applied the embedding to both sides in order to maximize the embedding capacity.

The watermark detection step is done during the video decoding, after the entropy decoding stage. If the value of the level is equal to p or p', it means that the watermark bit is "1", and if the value of the level is (p-1) or (p'+1), the watermark is equal to "0" as mentioned previously.

3.2 Video quality assessment

In order to assess the video quality after embedding the watermark, the objective quality is measured by the PSNR of the sequences. For subjective assessment, we have used a quality meter based on blockiness and blurriness detectors [6]. In this meter, FFT within a window of 32×32 pixels of the gradients of reference and processed images are taken. The added energy to the phase of the harmonics over the reference image is an indication of blockiness and the loss of energy in the amplitude of the harmonics would be that of blurriness [7]. Therefore by measuring the picture blockiness and blurriness one can gauge the video quality. Measurement results are normalized in the range 0 to 100 %. Scores higher than 60 % would be interpreted as a *Good* subjective quality in terms of blockiness or blurriness effects.

4. Results and analysis

In our experiments, the first 50 frames of the four CIF video clips: Walk, Coast Guard, Silent and Foreman (with Siemens logo) were coded at the quantizer parameter (Qp=28), at a frame rate of 30 pictures/s generating almost between 253 to 1139 kbits/ second. With the base-line profile, only the first frame was intra coded and the remaining frames were coded as P.

We suppose that the watermark was previously encrypted to ensure its security. For tests we used a pseudo random binary sequence which was inserted first in the I-frame then in the first 10 P frames and finally the embedding is applied in both I and P frames. The objective is to estimate the embedding capacity for CIF sequences and to evaluate the

impact of the watermark insertion on the video quality and the increase of the bitrate.

In a first step of our work, the embedding was done in coefficients +1 and – 1 because they offer the maximum embedding capacity as shown in the histogram of the quantized DCT coefficients of Walk sequence represented on figure 2. However, that introduces visible artifacts and a relatively high increase in bitrate. This is due to the fact that, in H.264 Levels ±1 play a crucial role in the encoding chain. Coefficients ± 2 have instead been chosen to embed the signature because they seem to offer the best compromise between the embedding capacity, the increase in bitrate and the video quality.

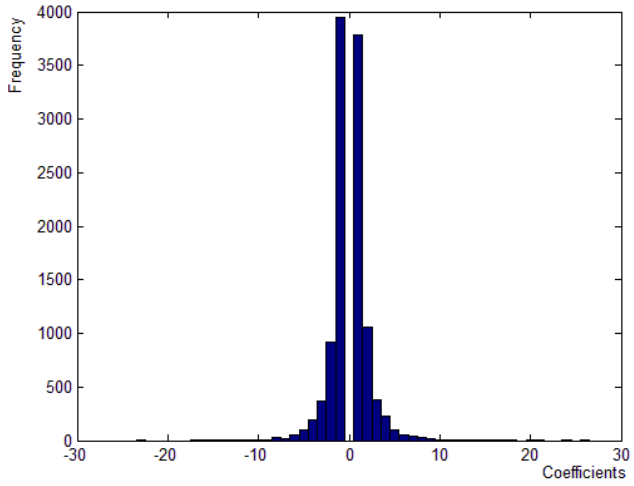


Figure 2. Histogram of the significant quantized DCT coefficients of I frame of Walk sequence.

The embedding was performed using both of the modes intra 16x16 and intra 4x4. Though, to improve the video quality for the sequences Silent and Foreman, tests have been redone, after experiments, by discarding the Intra 16x16 mode because of the distortion it may cause. Indeed, the number of the watermarked coefficients of a block depends on the picture content ; in some blocks many coefficients might be affected. Thus there is more chance that the distortion appears in intra 16x16 mode which is applied for homogeneous areas.

In the following points we will present results obtained for the embedding capacity, video quality and increase in bitrate. We will discuss the impact of the embedded signature on these two last parameters for the three following cases: Embedding the watermark in I frames, in P frames and in both of them.

4.1 Embedding capacity and Video quality

The embedding capacity depends on the picture content. In I frame, it varies between 1123 and 2925 bits corresponding to the CIF size sequences of Foreman and Coast Guard respectively.

Figures 3, 4 and 5 show the four unwatermarked I frames and the correspondent watermarked frames. An example of frame restoration is given in figure 3 (c) and an example of watermarked P frame is given in figure 5 (c).

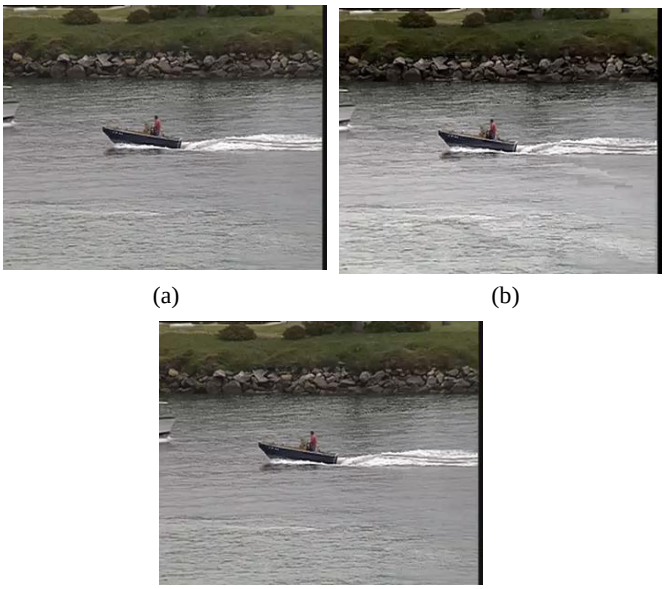


Figure 3 : (a) I frame of the unwatermarked Coast Guard sequence, (b) is the corresponding watermarked frame, (c) is the restored frame .

For the case of embedding the signature in I frame, the objective video quality, for all pictures, the inserted frame and the remaining ones, drops by almost between 12 to 13.5 dB (figure 6). These values along with the data hiding capacity and the increase in bitrate are tabulated in Tables 1, 2, 3 and 4.

Figure 4 : (a1), (b1) I frames of the unwatermarked sequences Walk and Silent respectively, (a2) and (b2) are the corresponding watermarked frames

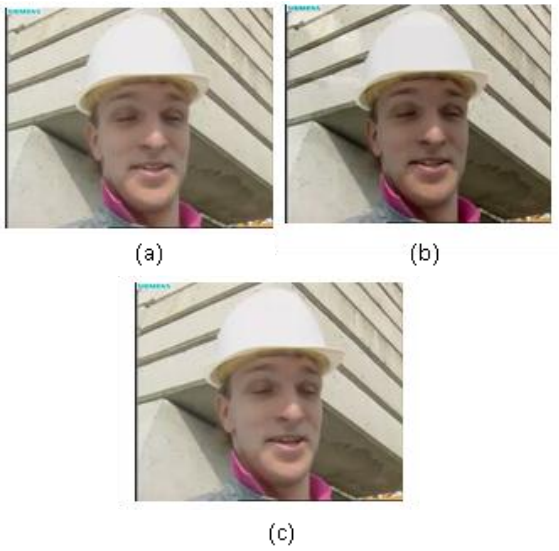
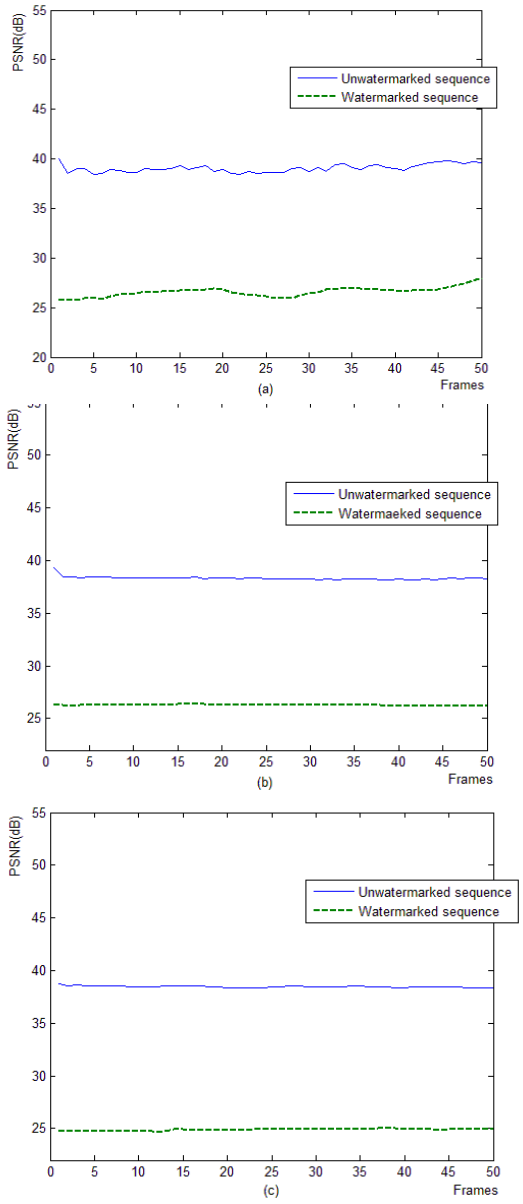
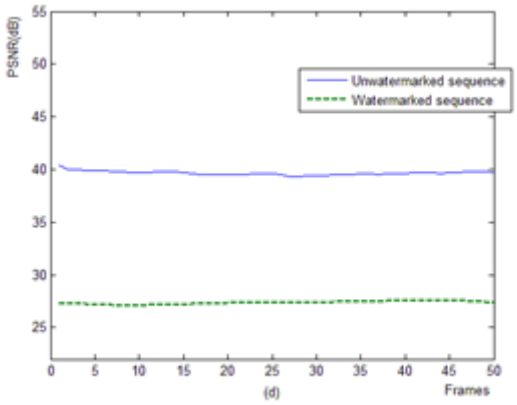


Figure 5: (a) I frames of the unwatermarked Foreman sequence, (b) corresponding watermarked frame, (c) Watermarked P frame (2nd frame).





Tables 1, 2, 3, and 4 also show the impact of the watermark on the subjective quality of the video, at various frames, (frame 1, F1 to frame 30, F30). The watermark is being embedded in the first frame, we are looking at the quality of I frame, the first P frame and three other P frames. Although the objective quality is deteriorated but subjective quality is maintained. The highest degradation is measured for Coast guard sequence which presents a blockiness of 72 % due to the high embedding capacity it offers (higher than the embedding capacity of Foreman I frame more than 2.5 times) but still visually may be acceptable as shown in figure 3 (b). The impact of this I watermark embedding on the rest of the frames is negligible in terms of blockiness or blurriness.

This can be explained by the fact that in shifted histogram technique the shift is generally applied to a relatively high number of coefficients. The effect of this shift is transmitted with the Inter prediction to P frames which reduces the PSNR for the whole sequence even if the watermark has not been embedded. Subjectively we can notice slight changes in the brightness of the frames (figure 4), but any possible degradation caused by the watermark embedding remains masked in the image texture.

For the case of embedding the watermark in P frames, instead of embedding the signature in one frame, it is spread over several frames since the embedding capacity in each P frame is very low comparing to I frame. Tables 1,2, 3 and 4 show that the embedding capacity for the four sequences varies between 168 and 5403 bits/10 P frames. The highest measured of this rate is observed for Walk and Coast Guard sequences which also present the highest encoding rates (1111.15 and 1139 kbits/ second respectively). The reduction in PSNR is relatively low, it varies - in accordance with the embedding capacity- between 0.02 and 1.28 dB corresponding to the sequences Silent and Walk respectively. Regarding the subjective quality, it is perfectly preserved, the highest degradation is shown for Walk sequence which presents a negligible blockiness of 95%, if we don't consider the blockiness effect already existing in the unwatermarked sequences. In addition to that, in our test the watermark embedding was done only in 10 P frames, the rest of the P frames of the GOP can also be exploited to hide more information without disturbing the video quality.

Table 1. The embedding capacity, the increase in bitrate and the video quality of Walk sequence

Video seq.	Embed. capacity (bits / fra.)	Incre. bitrate (%)	Red. PSNR (dB)	Blockiness %					Blurriness %				
				F1	F2	F10	F20	F30	F1	F2	F10	F20	F30
Unwatermarked Walk seq.	-	-	-	92	100	100	100	100	100	100	98	100	99
Walk seq. Watermark in I	1975	0.17	12.47	85	95	100	100	100	100	100	98	97	96
Walk seq. Watermark in 10 P	5403	0.42	1.28	92	95	98	99	100	100	100	100	100	100
Walk seq. Watermark in I & 10 P	7378	0.59	12.38	85	93	99	100	100	100	100	100	100	99

Table 2. The embedding capacity, the increase in bitrate and the video quality of Coast Guard sequence

Video seq.	Embed. capacity (bits / fra.)	Incre. bitrate (%)	Red. PSNR (dB)	Blockiness %					Blurriness %				
				F1	F2	F10	F20	F30	F1	F2	F10	F20	F30
Unwatermarked Coast Guard seq.	-	-	-	100	100	100	100	98	96	98	99	95	98
CG seq. Watermark in I	2925	0.19	12.04	72	78	86	90	88	100	100	100	100	100
CG seq. Watermark in 10 P	4327	0.21	0.81	100	100	100	100	98	98	100	100	98	99
CG seq. Watermark in I & 10 P	7252	0.41	12	72	85	83	87	86	100	100	100	100	100

Table 3. The embedding capacity, the increase in bitrate and the video quality of Silent sequence

Video seq.	Embed. capacity (bits / fra.)	Incre. bitrate (%)	Red. PSNR (dB)	Blockiness %					Blurriness %				
				F1	F2	F10	F20	F30	F1	F2	F10	F20	F30
Unwatermarked Silent seq.	-	-	-	96	95	100	96	100	99	100	99	98	98
Silent seq. Watermark in I	1791	0.56	13.5	89	90	93	91	94	100	100	98	97	97
Silent seq. Watermark in 10 P	168	0.04	0.02	96	95	99	96	100	99	100	99	98	98
Silent seq. Watermark in I & 10 P	1959	0.60	13.5	89	90	92	93	97	99	100	98	97	96

Table 4. The embedding capacity, the increase in bitrate and the video quality Foreman sequence

Video seq.	Embed. capacity (bits / fra.)	Incre. bitrate (%)	Red. PSNR (dB)	Blockiness %					Blurriness %				
				F1	F2	F10	F20	F30	F1	F2	F10	F20	F30
Unwatermarked Foreman seq.	-	-	-	89	92	100	100	100	100	100	100	100	99
Foreman seq. Watermark in I	1123	0.23	12.3	91	95	100	100	100	97	97	97	97	95
Foreman seq. Watermark in 10 P	839	0.19	0.41	89	92	100	100	100	100	100	100	100	99
Foreman seq. Watermark in I & 10 P	1962	0.43	12.25	91	95	100	100	100	97	97	97	97	95

Concerning the embedding performed in I and P frames, in terms of video quality results are close to those obtained for I frame embedding. Moreover, we can notice that, picture blurriness is preserved or even improved for Walk and Coast guard sequences. This is expected as the inserted data may act as dither in the image, creating artificial details to reduce picture blurriness for especially almost regular areas, which explain the fact that this is not observed when the signature is only embedded using the Inta 4x4 mode as presented in Table 3 and 4.

The embedding Capacity is obviously much more interesting since it is the addition of the two previous cases.

4.2 Increase in bitrate

Tables 1, 2,3 and 4 show that , the lowest and highest rates of the increase in bitrate are observed for Silent sequence. 0.04 % is the lowest rate corresponding to embedding the signature in P frames, and 0.6 % is the highest rate corresponding to embedding the signature in both I and P frames. However, to have a better idea on what represents that increase comparing to the embedding capacity, we calculated the rate between the embedding capacity and the increase in bitrate and we have noticed that for all the cases the embedding capacity is higher than the increase in bitrate introduced by the watermark insertion, the worse case in observed when the watermark is embedded in I for Walk sequence for which the bitrate increases by 1904.64 bits /second for an embedding capacity of 1975 bits and the best case when the watermark is embedded in P frames for Coast Guard sequent for which the bitrate increases by 2990.08 bits /second for an embedding capacity of 4327 bits. This is interesting insofar that rather than a second channel, data can be embedded and sent at lower bitrate.

Finally, the choice of I frame, P frames or both of them for the signature embedding depends on the application needs. Of course in case of the brightness modification introduced by the watermarking process is localized in only some areas of the watermarked frame, that may be noticeable as presented in figure 5 (b), it is thus preferable to choose only P frames to embed the signature and preserve the same video quality (figure 5 (c)). Though, in general, according to the tolerated increase in the bitrate and to the amount of information we need to carry in the video it is possible to choose between P frames that preserve the best the video quality, or in I frame because of its importance in the encoded sequence or in both of them so that the embedding capacity is maximized.

5. Conclusion

The shifted-histogram based method can be applied to the H264/AVC video standard. Indeed, despite the large differences between the objective quality (PSNR) of the watermarked and non-watermarked video sequences, their subjective quality differences are almost negligible.

For the four test video sequences, we noticed that the increase in bitrate varied according to the embedding frames type and number, but it didn't exceed the embedding capacity. This represents an advantage to carry information in the video sequence instead of a second channel.

Finally, According to the pictures content, the embedding capacity can reach interesting rates in I and P frames which can used separately or together according to the application demands in terms of embedding capacity, video quality and increase in bitrate.

References

[1] T.Weigand , G.Sullivan, G.Bjontegaard and A.Luthra, "Overview of the H.264/AVC video coding standard, " IEEE Trans. on Circuits and System for Video Technology, Vol.13, No.7, pp. 560-576, 2003.

[2] W-M. Chen, C-J. Lai and C-C. Chang, "H.264 Video Watermarking With Secret Image Sharing, " IEEE International Symposium on Broadband Multimedia Systems and Broadcasting (BMSB '09), Bilbao, Spain, 2009.

[3] M. Noorkami and R.M.Mersereau, "Compressed domain video watermarking for H264," IEEE Int. Conf. Image Processing, Genoa, Italy, 2005.

[4] Z. Ni, Y.Q. Shi, N. Ansari, and W. Su, "Reversible data hiding, " IEEE Transactions on Circuits and Systems for Video Technology, Vol.16, No.3 , pp.354-362, 2006.

[5] M.Fallahpour, D.Megias, M. Ghanbari, "High capacity, reversible data hiding in medical images," IEEE Int. Conf. on Image Processing (ICIP 2009), Cairo, Nov. 2009 (*to be published in IET Image Processing*, 2010).

[6] K.T.Tan, and M.Ghanbari, "Blockiness detection for MPEG2-coded video," IEEE Signal Processing Letters, Vol.**7,** No.8, pp. 213-215, 2000.

[7] I.P.Gunawan, and M.Ghanbari, "Reduced-reference video quality assessment using discriminative local harmonic strength with motion," IEEE Trans. on Circuits and Systems for Video Technology, IEEE-CSVT, Vol.18, No.1, pp. 71-73, 2008.

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Samira Bouchama has received the Electrical engineering degree from the National Polytechnic School (ENP) of Algiers (Algeria). She achieved the Master's degree from the same school (Signal and Communications Laboratory) in 2007. She is now working at the Research Center on Scientific and Technical Information (CERIST) of Algiers and pursuing as a doctoral student at ENP in the field of image and video watermarking.

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digital control systems. He is also registered as Professional Engineer from Pakistan Engineering Council.

Mohammed Ghanbari is a Professor of video Networking in the department of Computing and Electronic Systems, University of Essex, United Kigdom. He is best known for the pioneering work on two layer video coding of ATM networks, now is known as SNR scalability in the standard video codecs, which earned him the Fellowship of IEEE in 2001. He has registred for eleven international patents and published more than 400 technical papers on various aspects of video networking. He was the corecipient of A.H. Reeves prize for the best paper published in the 1995 proceeding of IEE in the theme of digital coding. He was also a coinvestigator of the European MOSAIC project studying the subjective assessment of picture quality, which resulted to ITU-R recommendation 500.

He is the co-author of Principles of Performance Engineering, book published by IEE press in 1997, and the author of video coding: an Introduction of standard codecs, book also published by IEE press in 1999, which received the Rayleigh prize as the best book of year 2000 by IEE. His latest book Standard Codecs: image compression to advanced video coding, was published by IEE 2003. He has been an organizing member of several international conferences and workshops. He was the general chair of 1997 international workshop on Packet Video and Guest Editor to 1997 IEEE transactions on circuits and systems for video Technology, Special issue on Multimedia technology and applications. He has served as Associate Editor to IEEE transactions on Multimedia and represented University of Essex as one of the six UK academic partners in the Virtual Centre of Excellence in Digital Broadcasting and Multimedia. He is a Fellow of IEEE, Fellow of IET and Charted Engineer (CEng).

Suppression of Random Valued Impulsive Noise using Adaptive Threshold

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***Abstract*:** In the proposed schemes adaptive threshold selection is emphasized for RVIN model. Incorporation of adaptive threshold into the noise detection process led to more reliable and more efficient detection of noise. Based on the noisy image characteristics and their statistics, threshold values are selected. Extensive simulations and comparisons are done with competent schemes. It is observed that the proposed scheme is better in suppressing impulsive noise at different noise ratios than their counterparts

***Keywords*:** RVIN: Random Valued Impulse Noise, SPN: Salt and Pepper Noise, PSNR: Peak Signal to Noise Ratio, PSP: Percentage of Spoiled Pixels.

1. Introduction

Noise Suppression from images is one of the most important concerns in digital image processing. ***Impulsive noise*** is one such noise, which may corrupt images during their acquisition, transmission and storage etc. A variety of techniques are reported to remove this type of noise. It is observed that techniques which follow the two stage process of detection of noise and filtering of noisy pixels achieve better performance than others. In this paper such schemes of impulsive noise detection and filtering thereof are proposed.

The models of impulsive noise considered in this paper are:

The first one is ***Salt & Pepper Noise (SPN) model***, where the noise value may be either the minimum or maximum of the dynamic gray scale range of the image.
The second one is ***Random Valued Impulsive Noise (RVIN) model***, where the noise pixel value is bounded by the range of the dynamic gray scale of the image.

The schemes used are:

The first scheme is based on ***second order difference of pixels*** in order to identify noisy pixels. The second scheme for SPN model uses ***fuzzy technique*** to locate contaminated pixels. The contaminated pixels are then subjected to median filtering. This detection–filtration is done recursively so that filtered pixels take part in the detection of noise in the next pixel.

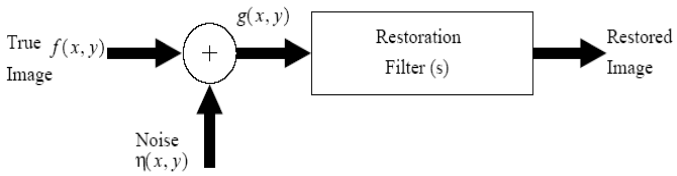

Fig1. Noise Removal Process

1.1 Filtering Techniques

Adaptive filtering has taken advantage of nonlinear filtering techniques. Non-adaptive nonlinear filters are usually optimized for a specific type of noise and signal. Adaptive filters become the natural choice and their performance depends on the accuracy of estimation of certain signal and noise statistics

1.1.1 Filtering without Detection

In this type of filtering a window mask is moved across the observed image. The mask is usually of size $(2N+1)^2$, where N is a positive integer. Generally the center element is the pixel of interest. When the mask is moved starting from the left-top corner of the image to the right-bottom corner, it performs some arithmetical operations without discriminating any pixel.

1.1.2 Detection followed by Filtering

This type of filtering involves two steps. In first step it identifies noisy pixels and in second step it filters those pixels. Here also a mask is moved across the image and some arithmetical operations is carried out to detect the noisy pixels. Then filtering operation is performed only on those pixels which are found to be noisy in the previous step, keeping the non-noisy intact.

1.1.3 Hybrid Filtering

In such filtering schemes, two or more filters are suggested to filter a corrupted location. The decision to apply a particular filter is based on the noise level at the test pixel location or performance of the filter on a filtering mask.

1.2 Impulsive Noise

Different types of noise frequently contaminate images. Impulsive noise is one such noise, which may affect images at the time of acquisition due to noisy sensors or at the time of transmission due to channel errors or in storage media due to faulty hardware. Two types of impulsive noise models are described below. Let $Y_{i,j}$ be the gray level of an original image Y at pixel location (i, j) and $[n_{min}, n_{max}]$ be the dynamic range of Y. Let $X_{i,j}$ be the gray level of the noisy image X at pixel (i, j) location. *Impulsive Noise* may then be defined as:

$$X_{i,j} = \begin{cases} Y_{i,j} & \text{with } 1-p \\ R_{i,j} & \text{with } p \end{cases} \quad (1)$$

where, $R_{i,j}$ is the substitute for the original gray scale value at the pixel location (i, j). When $R_{i,j} \in [n_{min}, n_{max}]$, the image is said to be corrupted with *Random Valued Impulsive Noise* (RVIN) and when $R_{i,j} \in \{n_{min}, n_{max}\}$, it known as *Fixed Valued Impulsive Noise* or *Salt & Pepper Noise* (SPN). Pixels replaced with RVIN and their surroundings exhibit very similar behavior. These pixels differ less in intensity, making identification of noise in RVIN case far more difficult than in SPN. The difference between SPN and RVIN may be best described by Figure 2.

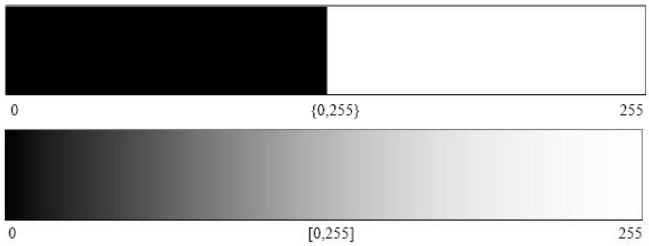

Fig.2. Representation of (a) *Salt & Pepper Noise* with $Ri,j \in \{n_{min}, n_{max}\}$,
(b) *Random Valued Impulsive Noise* with $Ri,j \in [n_{min}, n_{max}]$

In the case of SPN the pixel substitute in the form of noise may be either $n_{min}(0)$ or $n_{max}(255)$. Where as in RVIN situation it may range from n_{min} to n_{max}.

The metrics used for performance comparison of different filters (exists and proposed) are defined below.

a. Peak Signal to Noise Ratio (*PSNR*)

PSNR analysis uses a standard mathematical model to measure an objective difference between two images. It estimates the quality of a reconstructed image with respect to an original image. The basic idea is to compute a single number that reflects the quality of the reconstructed image. Reconstructed images with higher PSNR are judged better. Given an original image Y of size ($M \times N$) pixels and a reconstructed image $\hat{Y}$, the $PSNR(dB)$ is defined as:

$$PSNR(dB) = 10\log_{10}\left(\frac{255^2}{\frac{1}{M\times N}\sum_{i=1}^{M}\sum_{j=1}^{N}\left(Y_{i,j}-\hat{Y}_{i,j}\right)^2}\right) \quad (2)$$

b. Percentage of Spoiled Pixels (*PSP*)

PSP is a measure of percentage of non-noisy pixels changes their gray scale values in the reconstructed image. In other words it measures the efficiency of noise detectors. Hence, lower the *PSP* value better is the detection, in turn better is the filter performance

$$PSP = \frac{\text{number of non-noisy pixels changed their gray value}}{\text{total number of non-noisy pixels}} \times 100 \quad (3)$$

2. Efficient Impulsive Noise Removal Schemes

2.1 Decision Directed Median Filter (DDMF)

Usually the pixels located in the neighborhood of a test pixel are correlated to each other and they possess almost similar characteristics. Most of the reported impulse detection schemes exploit this feature of pixels. The scheme proposed here is one such novel technique of impulsive noise detection-suppression strategy from corrupted images. This scheme is simple but efficient and works alternatively in two phases: *detection of noisy pixels* followed by *median filtering [2]*.

2.1.1 Methodology

In a practical situation, since the probability p (1.5) is less than 1, all the pixels of a digital image are not corrupted with the impulsive noise. In addition, when the probability of corruption is not cent percent, it is expected that the noisy pixel be surrounded by at least some healthy pixels. However, this assumption is not true as the noise density becomes very high. In any case, the total number of corrupted pixels is less than the total number of pixels in the image. Hence, it is not required to perform filtering operation on every pixel for eliminating the impulsive noise. Rather, it is computationally economical to filter only the corrupted pixels leaving the healthy pixels unchanged. This approach reduces the blurring effect in the restored image, as the magnitude of healthy pixels is not affected by filtering. Basically, the noise removal method proposed in this paper constitutes two tasks: identification of corrupted pixels and filtering operation on those corrupted pixels. Thus the effectiveness of this scheme lies on the accuracy and robustness of detection of noisy pixels and efficiency of the filtering methodology employed. Many researchers have suggested [5][6][7] various methods for locating the distorted pixels as well as filtering techniques. Each of these methods has different shortcomings and hence fails to reproduce images very close to original ones. These are over-filtering distortion, blurring effect or high computational involvement. In addition, as the density of the impulsive noise is gradually increased, the quality of the image recovered by the existing methods correspondingly degrades. The scheme proposed here, is an improved impulsive noise detection scheme followed by recursive median filtering to overcome many of the shortcomings observed in the existing methods. To achieve this objective, it is necessary to devise an effective impulse detection scheme prior to filtering operation. The proposed scheme employs a second order difference based impulse detection mechanism at the location of a test pixel. The mathematical formulation of the proposed method is presented in (2).

$$\hat{Y}_{i,j} = \begin{cases} Z_{i,j} & \text{if } d_{i,j} = 0 \\ X_{i,j} & \text{if } d_{i,j} = 1 \end{cases} \quad (4)$$

where, di,j is the decision index that controls the filtering operation and estimates the filtered output $\hat{Y}i,j$ from the observed image Xi,j and filtered pixel value Zi,j. If the impulse detector determines that the center pixel of test window is noisy, then $di,j = 0$, otherwise $di,j = 1$. When $di,j = 0$, the corrupted pixel undergoes median filtering. On the other hand $di,j = 1$, the window is skipped and the process is repeated. Unlike in conventional methods, the filtering operation is performed selectively based on the decision of the impulse detector. Hence the proposed method is named as *Decision Directed Median Filter* (DDMF). The schematic diagram of the proposed filtering scheme is shown in Fig.3.

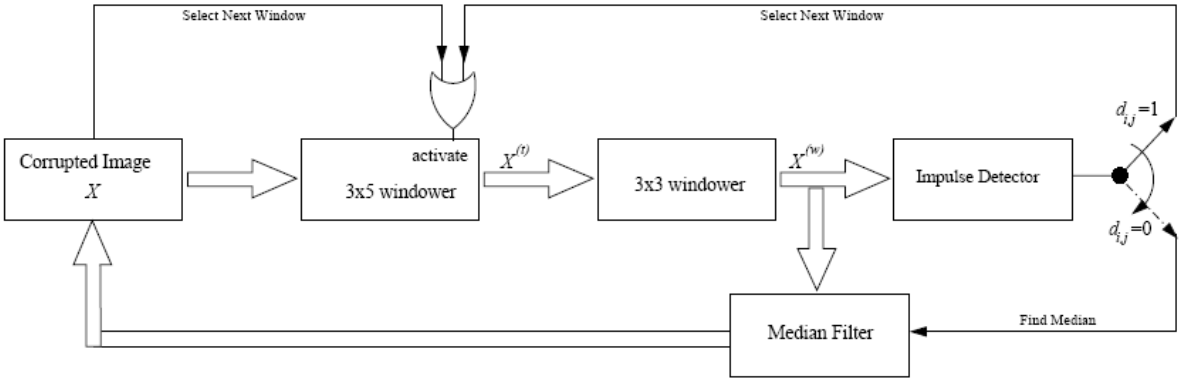

Figure 3: Schematic Diagram of the Decision Directed Median Filter

2.2 Recursive Median Filtering Algorithm

Based on the algorithm corrupted pixels are identified across the image. Then the filtering operation is carried out only on those distorted pixels. The recursive filtering operation computes the median value of a 3×3 window $X(w)$ surrounding the corrupted center pixel and substitutes this value at the location of the faulty pixel unlike the conventional median filter. In the next adjacent window, the healthiness of its center pixel is tested considering the gray level of the already filtered pixel rather than that of the original one. Mathematically,

$$\hat{Y}_{i,j} = \begin{cases} Y_{i,j} & \text{if } d_{i,j} = 1 \\ Z_{i,j} & \text{otherwise} \end{cases} \quad (5)$$

where,

$$Z_{i,j} = \text{median}\{X^{(w)}_{i-k,j-l}, (k,l) \in X^{(w)}\}$$

3. Fuzzy Impulsive Noise Detection

In this section, a fuzzy based filtering scheme namely *F*uzzy Impulsive Noise Detection (FIND) is proposed. It employs a fuzzy detection scheme to identify pixels corrupted with impulsive noise and subsequently filter the noisy pixels using recursive median filter. The detector is responsible for ascertaining the healthiness of a pixel in a test window by utilizing the gray level information in its neighborhood. The median filtering is applied to corrupted locations only leaving the non-corrupted ones intact. Such selective filtering operation prevents from edge jittering and blurring of images

3.1 Methodology

The proposed filtering scheme is a selective one and consists of two stages: decision making regarding the presence of impulsive noise (*salt & pepper*) at a test pixel location and median filtering only of corrupted pixels. Hence, detection operation is carried out at all locations but filtering is performed only at selected locations. From the corrupted image a 3 × 3 window is selected and a fuzzy detection is employed to derive a decision regarding the presence of impulse at the center pixel. Accordingly, the center pixel is replaced with the median value of the pixels in its neighborhood prior to the selection of the next window. The overall block diagram of the combined filter structure is depicted in Figure 4.

3.2 The FIND Algorithm

The impulse detection for the proposed filter is based on the fuzzy inference logic. In any fuzzy application, the challenge lies in Fuzzification and defuzzification process [11]. In our case, we use a triangular fuzzy membership function that utilizes only two linguistic variables. In the following, the proposed FIND algorithm is outlined stepwise.

i. Select the first test window of size 3 × 3 from the corrupted image *X*
ii. Convolve *Xw* with the two kernels to obtained Δ1 and Δ2
iii. Apply Mamdani fuzzy model for two-input one-output
iv. Compute the strength of each linguistic variable using triangular fuzzy membership functions
v. Evaluate the fuzzy rules using Zadeh logic for AND implication.
vi. Construct the consequent membership function as shown in Figure 2.5 from nine active rules for a system with two inputs and one output.
vii. Obtain a crisp value (*M*) from the fuzzy set (Defuzzification) by using center of-gravity method and apply to a decision process

$$d_{i,j} = \begin{cases} 0 & \text{if } M \text{ belongs to noisy class} \\ 1 & \text{otherwise} \end{cases}$$

viii. Invoke the median filtering on *Xi,j* using *Xw*,if *di,j* = 0 else go to step (viii).
ix. Shift the test window column wise and then row wise to cover the entire image pixels.
x. Repeat steps (ii) through (ix) for all windows.

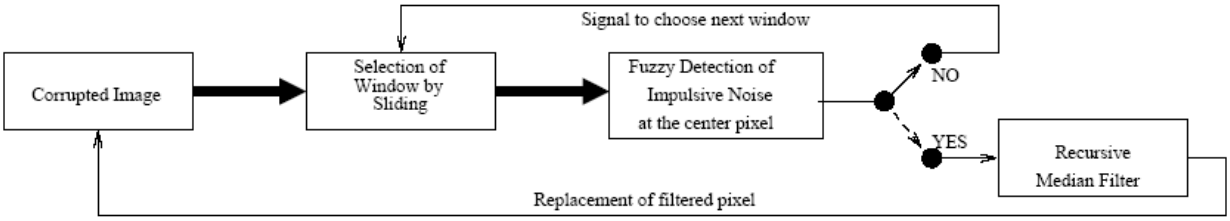

Figure 4: Block Diagram of the Proposed Filter

Table1.

	Leena	Girl	Clown	Gatlin
SM5X5	28.66	28.43	24.78	30.70
MED3X3	34.19	30.63	22.75	31.19
MED5X5	27.71	27.41	21.75	28.45
PnV	28.97	29.47	21.79	28.32
WMedk=1	29.87	28.63	23.47	28.43
WMedk=2	22.85	21.76	20.70	21.84
TMED	28.26	27.29	20.45	29.84
ATMED	28.03	27.05	20.13	29.75
DDMF	40.70	38.00	34.00	42.40
FIND	30.25	32.46	25.85	35.05

4. Simulations and Results

The proposed schemes in this paper are simulated and their performance is compared with some of the recently reported schemes. Median (MED(3×3)) and (MED(5×5)) [2], Switching-Median (SM(5×5)) [3], Weighted Median (WM(k =1)) and (WM(k = 2)) [8], Peak and Valley (P n V) [9], TMED and ATMED [4] [10] are the compared schemes. *Lena* image is corrupted with *Salt & Pepper Noise* with density ranging from 1% to 30%. These images are then subjected to filtering by the proposed schemes along with the above listed schemes. The PSNR (*dB*) and PSP thus obtained are plotted in Figures 5 and 6.

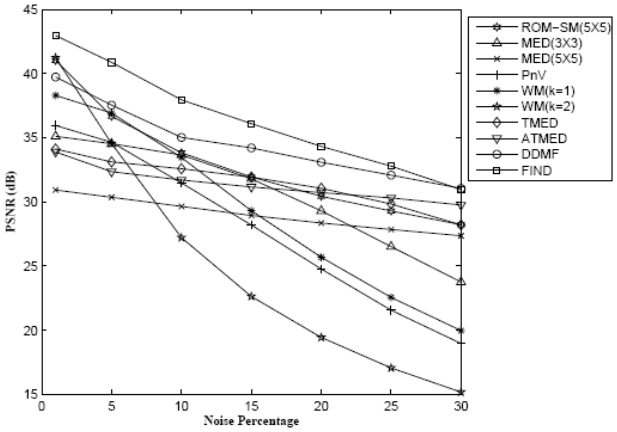

Figure 5 The PSNR (*dB*)

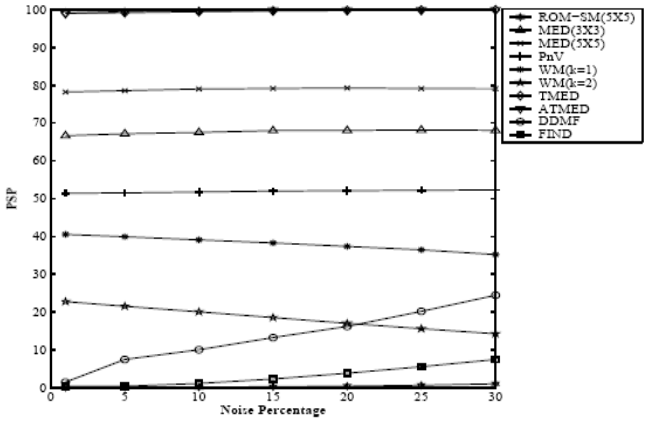

Figure 6 The PSP

Figure 7(a) is the true image of *Lena* and Figure 7(b) is the noisy version (20% SPN)

. Figures 7(c) to 7(l) shows the restored images

Impulsive Noise filtering of *Lena* image corrupted with 20% of SPN by different filters

.

5. Conclusion:

Two different noise suppression approaches are proposed in this chapter. Based on the second order difference the first scheme uses a threshold to determine impulses. In the fuzzy approach, two parameters are generated from the image. These parameters are then subjected to different stages of fuzzy techniques to identify the noise location. As can be seen from the plots both the proposed schemes out performs the existing schemes. One of limitations of these two techniques is that it use fixed value of threshold

References:

1. T. Chen and H. R. Wu. Adaptive Impulse Detection Using Center-Weighted Median Filters. *IEEE Signal Processing Letters*, 8(1):1 – 3, January 2001.
2. E. Abreu, M. Lightstone, S. K. Mitra, and K Arakawa. A New Efficient Approach for the Removal of Impulse Noise from Highly Corrupted Images. *IEEE Transactions on Image Processing*, 5(6):1012 – 1025, June 1996.
3. Z. Wang and D. Zhang. Progressive Switching Median Filter for the Removal of Impulse Noise from Highly Corrupted Images. *IEEE Transactions on Circuits and Systems–II: Analog and Digital Signal Processing*, 46(1):78 – 80, January 1999.
4. X. Xu and E. L. Miller. Adaptive Two-Pass Median Filter to Remove Impulsive Noise. In *Proceedings of International Conference on Image Processing 2002*, pages I–808 – I–811, September 2002.
5. K. Kondo, M. Haseyama, and H. Kitajima. An Accurate Noise Detector for Image Restoration. In *Proceedings of International Conference on Image Processing 2002*, volume 1, pages I–321 – I–324, September 2002.
6. C. Butakoff and I. Aizenberg. Effective Impulse Detector Based on Rank-Order Criteria. *IEEE Signal Processing Letters*, 11(3):363 – 366, March 2004.
7. R. Garnett, T. Huegerich, C. Chui, and W. He. A Universal Noise Removal Algorithm With an Impulse Detector. *IEEE Transactions on Image Processing*,14(11):1747 – 1754, November 2005.
[8] S. J. Ko and Y. H. Lee. Center Weighted Median Filters and Their Applications to Image Enhancement. *IEEE Transactions on Circuits and Systems*, 38(9):984 – 993, September 1991.
[9]P. S. Windyga. Fast Impulsive Noise Removal. *IEEE Transactions on Image Processing*, 10(1):173 – 179, January 2001.
[10] H. K. Kwan and Y. Cai. Fuzzy Filter for Image Filtering. In *Proceedings of Circuits and Systems, MWSCAS-2002, The 2002 45th Midwest Symposium*, volume 3, pages III–672 – III–675, August 2002.
[11] J. S. R. Jang, C. T. Sun, and E. Mizutani. *Neuro-Fuzzy and Soft Computing*. Prentice Hall International, USA, 1997.

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Paper Currency Recognition System using Characteristics Extraction and Negatively Correlated NN Ensemble

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***Abstract*: -** An efficient currency recognition system is vital for the automation in many sectors such as vending machine, rail way ticket counter, banking system, shopping mall, currency exchange service etc. The paper currency recognition is significant for a number of reasons. a) They become old early than coins; b) The possibility of joining broken currency is greater than that of coin currency; c) Coin currency is restricted to smaller range. This paper discusses a technique for paper currency recognition. Three characteristics of paper currencies are considered here including size, color and texture. By using image histogram, plenitude of different colors in a paper currency is calculated and compared with the one in the reference paper currency. The Markov chain concept has been considered to model texture of the paper currencies as a random process. The method discussed in this paper can be used for recognizing paper currencies from different countries. This paper also represents a currency recognition system using ensemble neural network (ENN). The individual neural networks in an ENN are skilled via negative correlation learning. The purpose of using negative correlation learning is to skill the individuals in an ensemble on different parts or portion of input patterns. The obtainable currencies in the market consist of new, old and noisy ones. It is sometime difficult for a system to identify these currencies; therefore a system that uses ENN to identify them is discussed. Ensemble network is much helpful for the categorization of different types of currency. It minimizes the chances of misclassification than a single network and ensemble network with independent training.

I. INTRODUCTION

By expansion of modern banking services, automatic schemes for paper currency recognition are significant in many applications. The requirements for an automatic banknote recognition system offered many researchers to build up a robust and dependable technique [1], [2], [3]. Speed and precision of processing are two vital factors in such systems. Of course, the precision may be much significant than the speed. Paper currency recognition systems should be clever to recognize banknotes from each side and each direction. Since banknotes may be faulty during circulation, the designed system should have an important precision in detecting torn or worn banknotes.

Artificial neural networks (ANN) have been applied in various application domains for solving real world problems such as, feature extraction from complex data sets, direct and parallel implementation of matching and search algorithm, forecasting and prediction in a rapidly changing environment, recognition and image processing applications etc. The currency recognition is one of the significant application domains of artificial neural networks. This paper discusses the ENN for currency recognition. NCL was used for the training of the network [7]. The use of NCL is to produce the diversity among the individual networks in ensemble. The final decision of the network is taken from voting among the individual NN. In voting each network gives a vote for a certain class and it is done by the winning neuron of that network.

II. LITERATURE REVIEW

Presently, there are a number of methods for paper currency recognition [1][2][3]. Using symmetrical masks has been used in [2] for recognizing paper currency in any direction. In this technique, the summation of non-masked pixel values in each banknote is evaluated and fed to a neural network for recognizing paper currency. In this technique, two sensors are used for recognition of the front and back of the paper currency, but the image of the front is the only criterion for decision. In another study for paper currency recognition [1], initially the edges of patterns on a paper currency are spotted. In the next step, paper currency is divided into N equal parts along vertical vector. Then, for each edge in these parts the number of pixels is added and fed to a three-layer, back propagation neural network. In this process, to conquer the problem of recognizing dirty worn banknotes, the following linear function is used as a pre-processor:

$$f(x) = F_a x + F_b \qquad (1)$$

where x is the given (input) image in gray scale, f(x) is the resultant image; and F_a, F_b and N are selected 3, -128 and 50 respectively [1]. In this technique, the algorithm depends to the number of paper currency denominations. Here, complexity of the system increases by increasing the number of classes. Therefore, this technique can be used only for recognition of a small number of banknote denominations. The technique discussed in this paper is not dependent to the number of paper currency classes. The features presented in this paper are independent to the way that a paper currency is placed in front of the sensor. It needs to be known that the discussed technique may not be able to differentiate genuine notes from counterfeits. Indeed, methods such as [8] which use infrared or ultraviolet spectra may be used for discriminating between genuine and counterfeits notes.

Most of paper currency recognition techniques use a single multilayer feed-forward NN for the recognition [9]-[13]. The features are first extracted from the image then it applied to network for training. These uses edge detection technique for feature extraction [9] and it reduces the network size. For new notes feature extraction from edge detection is simple. But for the noisy notes it is very difficult. Furthermore the currency recognition system should be highly consistent. If a network takes a false classification it will be not practical. So a single network is not reliable enough. Therefore ENN [14] is presented in this paper to solve this problem. The negative correlation learning was to generate different individual NN in the ensemble, so that entire ensemble learns the input pattern completely.

III. CHARACTERISTICS EXTRACTION

In the discussed method characteristics of paper currencies are employed that are used by people for differentiating different banknote denominations. Basically, at first instance, people may not pay attention to the details and exact characteristics of banknotes for their recognition, rather they consider the common characteristics of banknotes such as the size, the background color (the basic color), and texture present on the banknotes. In this method, these characteristics are used in a decision tree to differentiate between different banknote denominations [4].

A. Size

The first phase of recognition in the algorithm considers size of the banknote. The significant issue in considering this characteristic is that the edges of banknotes are generally worn and torn due to circulations. Hence, its size is reduced, or even is increased slightly in rejoining the torn banknotes. Hence, the size condition in the decision tree is presented as:

$$| x - x_0 | < d_x \;\&\; | y - y_0 | < d_y \qquad (2)$$

Where x_0 and y_0 are size of the testing paper currency, and x and y are size of the reference paper currency. In eq (2) d_x and d_y represent alteration in the vertical and horizontal directions.

B. Color

Although a variety of colors are used in each paper currency, people normally use one or more dominant color for distinguishing between different paper currencies. In this paper, image of the banknote is transformed to an image in gray scale [5]. Then the gray scale level is reduced to have a significant judgment about the background color. In this paper the banknote images are quantized to 52 levels in gray scale. Then histogram of the image is calculated to find the plentitude of different color in the banknote.

C. Texture

In most of the countries, the size and color spectrum of some banknotes are very similar to each other. Hence, these characteristics of banknotes from different countries may be too close to each other. So, these characteristics may not be adequate to easily differentiate between different banknotes. Consequently, template of the banknotes is considered in addition to the forgoing characteristics. For recognizing the template, Markov chain [6] concept is used in representing random phenomenon.

A random process $\{x_k, k = 0, 1, 2,...\}$ is called a Markov chain if the possibility value in state x_{n+1} depends on just the possible value in state x_n , that is:

$$P(x_{n+1} = \beta \mid x_n = \alpha, x_{n-1} = \alpha_{n-1}, \ldots\ldots\ldots, x_0 = \alpha_0) = P(x_{n+1} = \beta \mid x_n = \alpha) \qquad (3)$$

This possibility can be shown by P_{ij}. The state space of a Markov chain can be shown in a matrix that is:

$$P = \begin{bmatrix} p11 & p12 \cdots & p1n \\ \vdots & \ddots & \vdots \\ pn1 & pn2 \cdots & pnn \end{bmatrix} \qquad (4)$$

where n is the number of states in the chain.

IV. STEPS FOR PAPER CURRENCY RECOGNITION

The algorithm for the discussed paper currency recognition system is presented as follows:

i) Banknote Size (number of pixels on horizontal and vertical axes) is calculated. If its size satisfies equation (2), it is considered as a possible true banknote.

ii) The banknote image histogram is calculated using the bin number set as 52.

iii) The transition matrices (N_x and N_y) are calculated for the banknote, then, the main diagonal elements of the matrices (namely D_x and D_y) are taken out as a feature for distinguishing between different denominations.

iv) The paper currency under observation is assigned to a denomination class if the Euclidean distances between the main diagonal elements of its transition matrices (D_x and D_y) and the main diagonal elements of the corresponding matrices of the reference banknote (D_{Rx} and D_{Ry}) are smaller than a predefined value.

v) At the end, the computed histogram in stage ii is compared with the histogram of the winner class in stage iv. If the Euclidian distance between the two histograms is larger than the predefined value, the banknote is assigned to an unknown class.

V. AN APPROACH USING NEGATIVE CORRELATION LEARNING

After discussing three characteristics of paper currencies (including size, color and texture) for the paper currency recognition, paper also focuses on ENN [14] for the same purpose. ENN [14] is a learning paradigm where a set of finite number of neural networks is trained for the similar task. The resultant vectors are applied simultaneously in all the ensembles. The negative correlation learning is to generate the diversity among the individual networks using a penalty term. NCL is used for the training of the network [15]. The use of NCL is to produce the diversity among the individual networks in ensemble. The final decision of the network is taken from voting among the individual NN. In voting each network gives a vote for a certain class and it is done by the winning neuron of that network. The NCL can be found elsewhere [15] and in brief, can be described below.

Assume a training set S of size N.

S = {(x (l), d(l), x(2), d(2)),……..(x(N), d(N))}

Where x is the input vector and d is the desired result.
Consider approximating d by forming an ensemble whose result F(n) is the average in the component NN result $F_i(n)$

$$F(n) = \frac{1}{M} \sum_{i=1}^{M} Fi(n) \qquad (5)$$

Where M and n refer to the number of NN in ensemble and training pattern, respectively. The error function Ei of the network i in NCL is given by the following eq (6).

$$Ei = \frac{1}{N} \sum_{i=1}^{N} Ei(n) \qquad (6)$$

$$= \frac{1}{N} \sum_{i=1}^{N} \left(\frac{1}{2}\right) (Fi(n) - d(n))2 + \frac{1}{N} \sum_{i=1}^{N} \lambda \; pi(n \qquad (7)$$

Where $E_i(n)$ is the value of the error function of the network i for the n^{th} training pattern. The first term of (7) is the empirical risk function of the network i. In the second term, Pi is a correlation penalty function is given by eq (8).

$$Pi(n) = (Fi(n) - F(n)) \sum_{j=i}^{M} (Fj(n) - F(n)) \qquad (8)$$

The partial derivative of Ei(n) with respective to the output network i on the n^{th} training pattern is

$$\frac{\partial Ei(n)}{\partial Fi(N)} = Fi(n) - d(n) + \lambda \frac{\partial Pi(n}{\partial Fi(N}$$
$$= Fi(n) - d(n) + \lambda \sum_{j=i} (Fj(n) - F(n)$$
$$= Fj(n) - d(n) + \lambda (Fi(n) - F(n)$$
$$= (1-\lambda)(Fi(n) - d(n)) + \lambda (F(n) - d(n \qquad (9)$$

The NCL is a simple extension to the standard Back-propagation algorithm [8]. In fact, the only alteration that is needed is to compute an extra term of the form $\lambda (Fi(n) - F(n)$ for the i^{th} network. During the training process, the entire ensemble interacts with each other through their penalty terms in the error functions. Each network i minimizes not only the difference between Fi(n) and d(n) , but also the difference between F(n) & d(n). That is, negative correlation learning considers errors what all other networks have learned while training a network.

VI. IMAGE PREPROCESSING

To give an image to the network an image needs to be preprocessed. In the discussed system, the gray scale image is used. The RGB image is converted into gray scale (Black-White) image by scanner. To reduce the network size the gray scale image is compressed. Each pixel of the compressed image is applied as an input to the network. The pixel values are kept in the range of 0-1.

VII. CONCLUSIONS

This paper discussed a technique for recognizing paper currencies of different countries. The technique uses three characteristics of paper currencies including size, color, and template. In this method the system can be trained for a new denomination banknote by just introducing one intact example of the banknote to it. In addition the system may recognize the banknote on each side or any direction.
Paper also focuses recognition system using negatively correlated ensemble neural network. The Ensemble network has better performance for recognition than single network. For training the negative correlation learning is used. In negative correlation the entire networks are negatively correlated through the strength of penalty term. The entire ensembles interact with each other and each network has specialized for a particular portion of input vector. So when a noisy pattern is applied the network will be able to recognize as a whole.

REFERENCES

[1] E. H. Zhang, B. Jiang, J. H. Duan, Z. Z. Bian, “Research on Paper Currency Reconition by Neural Networks”, Proceedings of the 2nd Int. Conference on Machine Learning and Cybernetics, 2003.

[2] F. Takeda and T. Nishikage, “Multiple Kinds of Paper Currency Recognition using Neural Network and application for Euro Currency”, IEEE Int. Joint Conf. on Neural Networks, pp: 143-147, 2000.

[3] F. Takeda, T. Nishikage and Y. Vatsuwato, “Characteristics Extraction of Paper Currency using Symmetrical Masks Optimized by GA and Neurorecognition of Multi-national paper currency”, World congress on computational Intelligence, vol. 1, pp: 634-639, 1998.

[4] Rafael C. Gonzalez, Richard E. Woods, Steven L. Eddins "Digital image processing using Matlab" .Printed in 2008.

[5] C. H. Gladwin, Ethnographic Decision Tree Modeling, Sage Publications,1989.

[6] M. Iosifescu, Finite Markov Processes and Their Applications, Wiley, New York, NY, 1980.

[7] M. Kim, D. Kim and S. Lee, “Face recognition using the embedded HMM with second-order blockspecific observations” Pattern Recognition, vol. 36,no. 11, pp. 2723-273, 2003.

[8] A. Vila a, N. Ferrer b, J. Mantec´on c, D. Bret´on c, J.F. Garc´ıa “Development of a fast and nondestructive procedure for characterizing and distinguishing original and fake euro notes”, Analytica Chimica Act,. no. 559, pp. 257–263, 2006.

[9] D. A. K. S. Gunaratna, N. D. Kodikara and H. L. Premaratne, "ANN Based Currency Recognition System using Compressed Gray Scale and Application for Sri Lankan Currency Notes", Proceedings of world academy of

science,engineering and technology, vol. 35, Nov. 2008, ISSN 2070-3740, pp.235-240.

[10] E. Zhang, B. Jiang, J. Duan and Z. Bian, "Research on paper currency recognition by neural networks" in Proc. 2nd International Conf. Machine Learning and Cybernetics, Xi'an, 2003, pp2193-2196.

[11] S. Omatu, T. Fujinaka, T. Kosaka, H. Yanagimoto, and M. Yoshioka. "Italian lira classification by lvq". In Proc. International Joint Conference on Neural Networks, IJCNN, pp 2947-2951, 2001,

[12] F. Takeda and T. Nishikage, "Multiple kinds of paper currency recognition using neural Network and application for euro currency". In Proc. IEEE International Joint Conference on Neural Networks, 2000, pp 143-147.

[13] M. Gori, A. Frosini and P. Priami. "A neural network-based model for paper currency

recognition and verification", IEEE Trans. Neural Networks, Nov.1996, ppI482-1490.

[14] L.K Hansel and P. Salamon, "Neural networks ensemble", IEEE Trans. Pattern Anal. Mach.

Intell., vol.12, no.l0, 1990, pp.993-1001.

[15] Yong Liu and Xin Yao "Ensemble Learning via Negative Correlation" Neural Networks Vol. 12, pp 1399-1404, 1999.

An Efficient Dictionary Based Compression and Decompression Technique for Fast and Secure Data Transmission.

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Abstract*: Compression algorithms reduce the redundancy in data representation to decrease the storage required for that data. Data compression offers an attractive approach to reducing communication costs by using available bandwidth effectively. Dictionary-based code compression techniques are popular as they offer both good compression ratio and fast decompression scheme. State of the art lossless data compressors are very efficient. While it is not possible to prove that they always achieve their theoretical limit (i.e. the source entropy), their effective performances for specific data types are often very close to this limit. Lossless compression researchers have developed highly sophisticated approaches, such as Huffman encoding, arithmetic encoding, the Lempel-Ziv family, Dynamic Markov Compression (DMC), Prediction by Partial Matching (PPM), and Burrows-Wheeler Transform (BWT) based algorithms. However, none of these methods has been able to reach the theoretical best-case compression ratio consistently, which suggests that better improved method is needed. The Burrows-Wheeler Transform, or BWT, transforms a block of data into a format that is extremely well suited for compression. The block sorting algorithm they developed works by applying a reversible transformation to a block of input text. The transformation does not itself compress the data, but reorders it to make it easy to compress with simple algorithms such as move to front encoding[1]. The basic strategy adopted in this paper is to preprocess the text and transform it into some intermediate form which can be compressed with better efficiency and which exploits the natural redundancy of the language in making the transformation. A technique called efficient Dictionary Based encoding is used to achieve this. It preprocesses the standards text db prior to conventional compression to improve the compression efficiency much better and provides security.*

Keywords*: Compression, decompression,star encoding, Efficient dictionary based compression and decompression,BWT*

1. Introduction

1.1 Compression

Data compression, in context is the science(the art) to represent information in a compact form. It is the process of converting an input data stream (the source stream or the original raw data) into another data stream (the output, or the compressed, stream) that has a smaller size[2]. A stream is either a file or a buffer in memory. Data compression is popular for two reasons: (1) People like to accumulate data and hate to throw anything away. No matter how big a storage device one has, sooner or later it is going to overflow. Data compression seems useful because it delays this inevitability. (2) People hate to wait a long time for data transfers. When sitting at the computer, waiting for a Web page to come in or for a file to download, we naturally feel that anything longer than a few seconds is a long time to wait[8]. There are two major families of compression techniques when considering the possibility of reconstructing exactly the original source. They are called lossless and lossy compression. Certain compression methods are lossy. They achieve better compression by losing some information. When the compressed stream is decompressed, the result is not identical to the original data stream. Such a method makes sense especially in compressing images, movies, or sounds. If the loss of data is small, we may not be able to tell the difference. In contrast, text files, especially files containing computer programs, may become worthless if even one bit gets modified. Such files should be compressed only by a lossless compression method. There are many known methods for data compression. They are based on different ideas, are suitable for different types of data, and produce different results, but they are all based on the same principle, namely they compress data by removing redundancy from the original data in the source file. Data compression also offers an attractive approach to reduce the communication cost by effectively utilizing the available bandwidth in the data links[8]. Use of compression for storing text files has become inherent part of personal as well as commercial computing. The various compression applications available perform two functions, compression and decompression. The text document is first compressed and then the entire document is decompressed when required.

1.2 Decompression

Any compression algorithm will not work unless a means of decompression is also provided due to the nature of data compression. When compression techniques are discussed in general, the word compression alone actually implies the context of both compression and decompression. In many practical cases, the efficiency of the decompression algorithm is of more concern than that of the compression algorithm. For example, movies, photos, and audio data are often compressed once by the artist and then the same version of the compressed files is decompressed many times by millions of viewers or listeners [2].

2. Related work

Various sophisticated algorithms have been proposed for lossless text compression A very promising development in the field of lossless data compression is the Burrows-Wheeler Compression Algorithm (BWCA), introduced in 1994 by Michael Burrows and David Wheeler. The algorithm received considerable attention since of its Lempel-Ziv like execution speed and its compression performance close to state-of-the-art PPM algorithms.

A preprocessing method is performed on the source text before applying an existing compression algorithm. The transformation is designed to make it easier to compress the source file. The star encoding is generally used for this type of pre processing transformation of the source text. Star-encoding works by creating a large dictionary of commonly used words expected in the input files. The dictionary must be prepared in advance, and must be known to the compressor and decompressor. Each word in the dictionary has a star-encoded equivalent, in which as many letters a possible are replaced by the '*' character. For example, a commonly used word such the might be replaced by the string t**. The star-encoding transform simply replaces every occurrence of the word the in the input file with t**. If done properly, this means that transformed file will have a huge number of '*' characters. [5].This ought to make the transformed file more compressible than the original plain text. The existing star encoding does not provide any compression as such but provide the input text a better compressible format for a later stage compressor. The star encoding is very much weak and vulnerable to attacks. The encoded data has exactly the same number of characters, but is dominated by stars

3. Proposed work and objectives:

The goal of this project is to develop new transformations for lossless text compression to incorporate fast, secure and to achieve a good compression ratio in transmissions .The approach consists of exploiting the natural redundancy of the English language by encoding text into some intermediate form before applying the backend compression algorithm which increases the context for compression. The encoding scheme uses dictionaries to co-relate words in the text and the transformed words. At the receiver end the file is decompress by using the same backend algorithm and the intermediate form is converted to text by the use of dictionary.

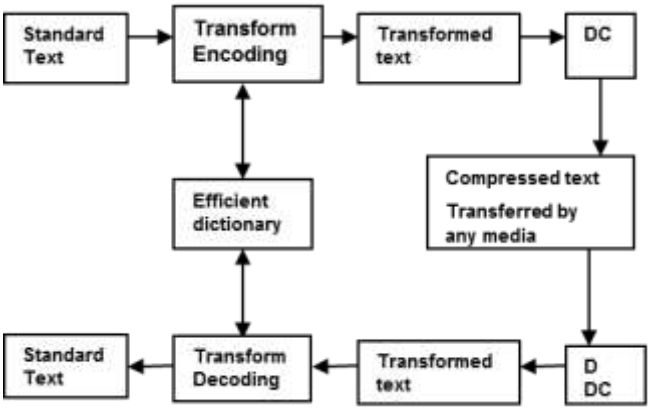

Figure 2 Block diagram

3.1 Formulation of Dictionary

The heart of dictionary coding is the formulation of the dictionary. A successfully built dictionary results in data compression; the opposite case may lead to data expansion. According to the ways in which dictionaries are constructed, dictionary coding techniques can be classified as static or adaptive. At present, dictionary-based compression schemes using static dictionaries are mostly ad hoc, implementation dependent and not general purpose. Most well-known dictionary algorithms are adaptive. Instead of having a completely defined dictionary when compression begins, adaptive schemes start out either with no dictionary or with a default baseline dictionary. As compression proceeds, the algorithms add new phrases to be used later as encoded tokens. The basic principle behind adaptive dictionary programs is relatively easy to follow.

3.2 Transformed Encoding

Once have dictionaries, Need to examine the input text and find a string of symbols that matches an item in the dictionary. Then the index of the item to the dictionary is encoded. This process of segmenting the input text into disjoint strings (whose union equals the input text) for coding is referred to as parsing. Obviously, the way to segment the input text into strings is not unique. Parsing strategy which is used is a greedy parsing. With greedy parsing, the encoder searches for the longest string of symbols in the input that matches an item in the dictionary at each coding step. Greedy parsing may not be optimal, but it is simple in implementation.

Algorithm:

```
Dictionary = empty; Prefix = empty;
 DictionaryIndex = 1;
while(characterStream is not empty)
{Char = next character in characterStream;
     if(Prefix + Char exists in the Dictionary)
      then Prefix = Prefix + Char ;
      else
       {   if(Prefix is empty)
                 CodeWordForPrefix= 0 ;
           else
```

```
            CodeWordForPrefix = DictionaryIndex for
            Prefix ;
          Output: (CodeWordForPrefix, Char) ;
          insertInDictionary( ( DictionaryIndex , Prefix +
Char) );
                DictionaryIndex++ ;
                Prefix= empty ;
             }
}
if(Prefix is not empty)
{
          CodeWordForPrefix = DictionaryIndex for Prefix;
    Output: (CodeWordForPrefix,   ) ;
}
```

Example: Suppose the codeword are indexed starting from 1:Inputed string ABBCBCABABCAABCAAB
Compressed string (code words):
(0, A) (0, B) (2, C) (3, A) (2, A) (4, A) (6, B)
1 2 3 4 5 6 7 Index

Each code word consists of an integer and a character where character is represented by 8 bits. The number of bits n required to represent the integer part of the codeword with index i is given by:

$$n = \begin{cases} 1 & \text{if } i = 1 \\ \lceil \log_2 i \rceil & \text{if } i > 1 \end{cases}$$

Number of Bits required is (1 + 8) + (1 + 8) + (2 + 8) + (2 + 8) + (3 + 8) + (3 + 8) + (3 + 8) = 71 bits

3.3 Compression and decompression using Backend algorithm.

Any efficient backend algo(PPM/BWT) can be used to compress as well as decompress the data.The lossless Burrows-Wheeler compression algorithm has received considerable attention over recent years for both its simplicity and effectiveness. It is based on a permutation of the input Sequence the Burrows-Wheeler transformation which groups symbols with a similar context close together. The BWT is performed on an entire block of data at once. Most of today's familiar lossless compression algorithms operate in streaming mode, reading a single byte or a few bytes at a time. But with this new transform, we want to operate on the largest chunks of data possible. Since the BWT operates on data in memory, you may encounter files too big to process in one fell swoop. In these cases, the file must be split up and processed a block at a time. A typical scheme of the Burrows-Wheeler Compression Algorithm (BWCA) is presented and consists of four stages.

I/p->BWT→MTF->RLE->Huffman->O/p

Decompression

o/p<-BWT<-MTF<-RLE<-Huffman<-I/p

Each stage is a block transformation of the input buffer data and forwards the output buffer data to the next stage. The stages are processed sequentially from left to right for compression; for decompression they are processed from right to left with the respective backward transformations. For compression, the first stage is the BWT. As the first stage of the compression algorithm.The purpose of this stage is the reordering of the input data depending on its context. The reordering produces many runs of equal symbols inside the output data of the BWT stage. In order to move the reordered symbols back into their original positions, a backward transformation exists, which reproduces exactly the input sequence. [1]

3.4. Decoding

Decoding (decompressor) in dictionary method is very simple. It will not parse the given input string. It will use the same dictionary and will decode the sequence.

4. Performance Analysis:

The performance issues such as Bits Per Character (BPC) and conversion time are compared for the three cases i.e., simple BWT, BWT with Star encoding and BWT with our proposed Efficient Dictionary method The results are shown graphically which proves that the proposed method has a good compression as well as reconstruction (Decompression) time.

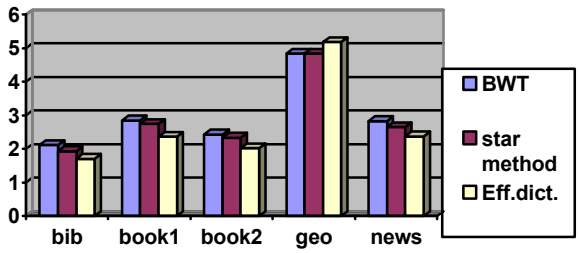

Figure 3: Encoding

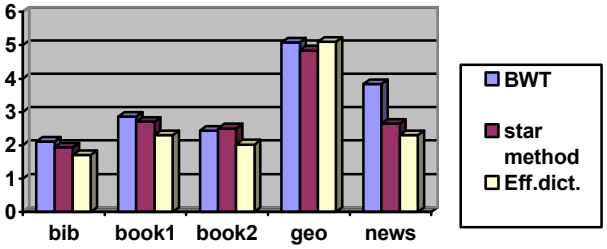

Figure 4: Decoding

5. Conclusion

An efficient Dictionary-based compression techniques use no statistical models. They focus on the memory on the strings already seen. The compression algorithm and the decompression algorithm build an identical dictionary independently. The advantage of this is that the compression algorithm does not have to pass the dictionary to the

decompressor.An efficient scheme is an excellent improvement in text data compression and added levels of security over the existing methods. This method will maintain the compression ratio with a good reconstruction at the receiver's end.

6. References

[1] Burrows, M, Wheeler, D. A Block-Sorting Lossless Data Compression Algorithm. Technical report, Digital Equipment Corporation, Palo Alto, California, 1994, URL (March 2006):
http://citeseer.ist.psu.edu/76182.html..

[2] E-book on Fundamentals of data compression by ida mengui pu.

[3] Isal, R, Moffat, A, Ngai, A. Enhanced Word-Based Block-Sorting Text Compression.In Proceedings of the twenty-fifth Australasian conference on Computer science, Volume 4,January 2002, 129□138, 2002.

[4] [KrMu96] H. Kruse and A. Mukherjee. .Data Compression Using Text Encryption. *Proc. Data Compression Conference*, 1997, IEEE Computer Society Press,1997.

[5] [KrMu97] H. Kruse and A. Mukherjee. .Preprocessing Text to Improve Compression, IEEE Computer Society Press,1997, p. 556.

[6] [Welc84] T. Welch, .A Technique for High-Performance Data Compression.,*IEEE Computer*, Vol. 17, No. 6, 1984.

[7] [ZiLe77] J. Ziv and A. Lempel. .A Universal Algorithm for Sequential Data Compression., *IEEE Trans. Information Theory*, IT-23, pp.237-243.

[8] E-Book A complete Reference on data Compression by David Solomon.

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Effective Compression Technique by Using Adaptive Huffman Coding Algorithm for Xml Database

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Abstract*: The Extensible Markup Language (XML) is one of the most important formats for data interchange on the Internet. XML documents are used for data exchange and to store large amount of data over the web. These documents are extremely verbose and require specific compression for efficient transformation. In this proposed work we are enhancing the existing compressors which uses Adaptive Huffman coding. It is based on the principle of extracting data from the document, and grouping it based on semantics. The document is encoded as a sequence of integers, while the data grouping is based on XML tags/attributes/comments. The main disadvantage of using XML documents is their large sizes caused by highly repetitive (sub) structures of those documents and often long tag and attribute names. Therefore, a need to compress XML, both efficiently and conveniently to use. The re-organized data is now compressed by adaptive Huffman coding. The special feature of adaptive Huffman coding algorithm is that, it has extremely accurate compression as well as it eliminates the repetition of dictionary based words in xml database. Using Adaptive Huffman algorithm, we derived probabilities which dynamically changed with the incoming data, through Binary tree construction.*

Keywords*: Compression, decompression, , Efficient XML compression and decompression,Adaptive Huffman coding.*

1. Introduction

The Extensible Markup Language (XML) is one of the most important formats for data interchange on the Internet. XML documents are used for data exchange and to store large amount of data over the web. These documents are extremely verbose and require specific compression for efficient transformation. In this proposed work we are enhancing the existing compressors which uses Adaptive Huffman coding. It is based on the principle of extracting data from the document, and grouping it based on semantics[1].The document is encoded as a sequence of integers, while the data grouping is based on XML tags/attributes/comments. The main disadvantage of using XML documents is their large sizes caused by highly repetitive (sub) structures of those documents and often long tag and attribute names. Therefore, a need to compress XML, both efficiently and conveniently to use. The design goal of Effective compression of XML database by using Adaptive Huffman coding is to provide extremely efficient and highly accurate compression of XML documents while supporting "online" usage. In this context, "online" usage means: (a) only one pass through the document is required to compress it, (b) compressed data is sent to the output stream incrementally as the document is read, and (c) decompression can begin as soon as compressed data is available to the decompressor. Thus transmission of a document over a heterogeneous systems can begin as soon as the compressor produces its first output, and, consequently, the decompress or can start decompression shortly thereafter, resulting in a compression scheme that is well suited for transmission of XML documents over a wide-area network.

2. Related work

Various sophisticated algorithms have been proposed for lossless text compression. A very promising development in the field of lossless data compression is the Burrows-Wheeler Compression Algorithm (BWCA), introduced in 1994 by Michael Burrows and David Wheeler. The algorithm received considerable attention since of its Lempel-Ziv like execution speed and its compression performance close to state-of-the-art PPM algorithms. A preprocessing method is performed on the source text before applying an existing compression algorithm. The transformation is designed to make it easier to compress the source file. The star encoding is generally used for this type of preprocessing transformation of the source text. Star-encoding works by creating a large dictionary of commonly used words expected in the input files. The dictionary must be prepared in advance, and must be known to the compressor and decompressor.

Several proposals and references there in make use of the observation that the pioneering work in this domain was XGRind which was based on static Huffman coding. XGRIND was the first XML-conscious compression scheme to support querying without full decompression.[7] Element and attribute names are encoded using a byte-based scheme, and character data is compressed using static Huffman coding. Use of the latter technique significantly slows down the compression process, since two passes over the original document are required (first to gather probability data for the

compression model, and a second time to perform the encoding according to the generated model). XPRESS also supports querying of compressed data and claims to achieve better compression than XGRIND. [8] However, it uses a semi-adaptive form of arithmetic coding which also necessitates two passes Over the original XML document.

3. Proposed work and objectives:

3.1 Compression Techniques for XML Database

- **Lossless Compression**

Lossless compression techniques provide exact recovery of the original data from their compressed version. Any information contained in an original cannot be lost during compression and reconstruction. These techniques are used widely in applications to save storage space and network bandwidth. Since an XML compressor needs to preserve all data content, only lossless compression techniques can be used.

3.1.1Huffman coding

In computer science and information theory, Huffman coding is an entropy encoding algorithm used for lossless data compression. The term refers to the use of a variable-length code table for encoding a source symbol (such as a character in a file) where the variable-length code table has been derived in a particular way based on the estimated probability of occurrence for each possible value of the source symbol. It was developed by David A. Huffman "A Method for the Construction of Minimum-Redundancy Codes.Huffman coding uses a specific method for choosing the representation for each symbol, resulting in a prefix code (sometimes called "prefix-free codes", that is, the bit string representing some particular symbol is never a prefix of the bit string representing any other symbol) that expresses the most common source symbols using shorter strings of bits than are used for less common source symbols.

In this proposed work we are enhancing the existing compressors which uses Adaptive Huffman coding
Adaptive Huffman coding for xml database compression

XML simplifies data exchange among heterogeneous computers, but it is notoriously verbose and has spawned the development of many XML-specific compressors and binary formats"[10]. We can present an XML test and a combined efficiency metric integrating compression ratio and execution speed. With the help of adaptive Huffman compression technique. The Adaptive Huffman compression is more efficient than static Huffman compression it is an important dimension for lossless data compression. In computer science and information theory Huffman coding is an entropy encoding algorithm used for lossless data compression.
Basically, the Adaptive algorithm states that a received symbol's node (*current node*) must be promoted (via the *Update* method) as the *highest* numbered node among nodes that are of *equal* weight. Simply swap the two nodes (i.e., the highest numbered node and the *current* node which must now become the *new* highest numbered node) and the tree maintains the Sibling property, as well as ensuring a correct tree. The parent of the current node then becomes the *new* current node, and the process continues until the root of the tree is reached

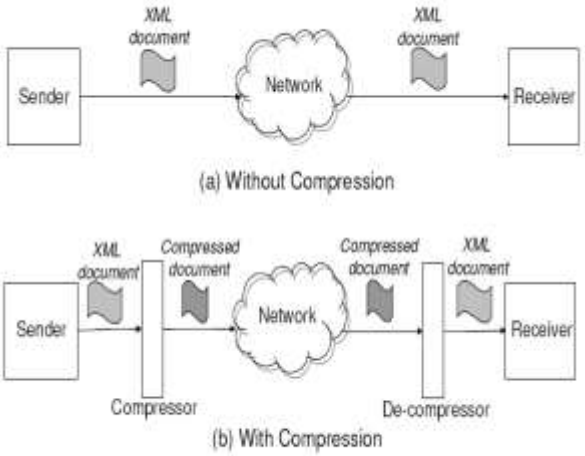

Figure 1.1: Transferring XML Document on the Network

3.1.2Adaptive Huffman Compression Technique

The proposal here is that to design an efficient way of compressing XML documents by using adaptive Huffman coding. High compression ratio and speed is equally important. We also require the transformation of xml database to be fully reversible and decompress able so that the decompressed document is a mirror image of the original. Huffman coding requires prior knowledge of the probabilities of the source sequence. If this knowledge is not available, Huffman coding becomes a two pass procedure: the statistics are collected in the first pass and the source is Encoded in the second pass. In the Adaptive Huffman coding procedure, neither transmitter nor receiver knows anything about the statistics of the source sequence at the start of transmission. Using Adaptive Huffman algorithm, we derived probabilities which dynamically changed with the incoming data, through Binary tree construction. Thus the Adaptive Huffman algorithm provides effective compression by just transmitting the node position in the tree without transmitting the entire code. Unlike static Huffman algorithm the statistics of the sensor data need not be known for encoding the data.Thats why adaptive Huffman is extremely accurate with respect to the compression.

The design goal of Effective compression of XML database by using Adaptive Huffman coding is to provide extremely efficient and highly accurate compression of XML documents while supporting "online" usage. In this context, "online" usage means: (a) only one pass through the document is required to compress it, (b) compressed data is sent to the output stream incrementally as the document is read, and (c) decompression can begin as soon as compressed data is available to the decompressor. Thus transmission of a document over a heterogeneous systems can begin as soon as the compressor produces its first output, and, consequently, the decompress or can start decompression shortly there after, resulting in a compression scheme that is well suited for transmission of XML documents over a wide-area network. Compression Performance ,Compression Time (CT) and decompression time. The

- **Objectives of proposed work are:**

To provide an Effective compression technique by using Adaptive Huffman coding algorithm

- **Adaptive Huffman coding for xml database compression**

XML simplifies data exchange among heterogeneous computers, but it is notoriously verbose and has spawned the development of many XML-specific compressors and binary formats. We can present an XML test and a combined efficiency metric integrating compression ratio and execution speed. With the help of adaptive Huffman compression technique. The Adaptive Huffman compression is more efficient than static Huffman compression it is an important dimension for lossless data compression. In computer science and information theory Huffman coding is an entropy encoding algorithm used for lossless data compression.

- **The Sibling Property of Adaptive Huffman Algorithm**

In dynamic coding, it is not enough to just have a symbol tree: the tree must be a "correct" *Huffman* tree. Thus, the tree is recomputed or *updated* to ensure a correct tree. Re computation of the tree is done dynamically, and the decoder also maintains the same tree that the encoder creates.

Adaptive Huffman Algorithm maintains a property to create a compact tree as much as possible. This is called the *Sibling Property*. Accordingly, the Sibling property ensures a *Huffman* tree in the fastest manner as possible; re computation of the tree always maintains the Sibling property.

The Sibling Property defines a binary tree to be a Huffman tree if and only if:

- all leaf nodes have non-negative weights (i.e., a leaf node can have a 0 weight), all internal nodes have exactly two children, and the weight of each parent node is the sum of its children's weights; and
- the nodes are numbered in increasing order by non-decreasing weight so that siblings are assigned consecutive numbers or *rank*, and most importantly, their parent node must be higher in the numbering [Vitter 1987].

With the Sibling property, nodes are *promoted* up the tree when necessary to assign them a minimal number of bits; that is, if they are gaining weight, they are assigned shorter bit codes. Some nodes may go down according to the statistics of the source; if one node goes up, there is certainly one node which goes down because the two nodes are simply swapped in the tree positions. Node promotion involves *constant* swapping of nodes.

Basically, the Adaptive algorithm states that a received symbol's node (*current node*) must be promoted (via the *Update* method) as the *highest* numbered node among nodes that are of *equal* weight. Simply swap the two nodes (i.e., the highest numbered node and the *current* node which must now become the *new* highest numbered node) and the tree maintains the Sibling property, as well as ensuring a correct tree. The parent of the current node then becomes the *new* current node, and the process continues until the root of the tree is reached.

- **Why Compress XML**
  - XML is verbose
  - XML documents has repetitive structures of data
  - Each non-empty element tag must end with a matching closing tag -- <tag>data</tag>
  - Ordering of tags is often repeated in a document (e.g. multiple records)
  - Tag names are often long

4. Conclusion

Adaptive Huffman coding encodes an alphabet with ⁻fixed codes. That allows us to directly search keywords in the compressed data[6]. Since the data volumes reduced, such compressing of xml data may be even faster than the original data. The resulting output of proposed work will be that xml document compresses with the help of adaptive Huffman algorithm and that compressed data will be decompressed as well and provide original xml document over a heterogeneous systems.

References

[1] Bray, et al. "Extensible Markup Language (XML) 1.0", October 2000, http://www.w3.org/TR/REC-xml.

[2]G. Girardot and N. Sundaresam,:an encoding format for efficient representation and exchange of XML over the Web",*http://www9.org/w9cdrom/154/154.html*

[3]D. Huffman, "A Method for Construction ofMinimum-Redundancy Codes", *Proc. of IRE*, September 1952.

[4] H. Liefke and D. Suciu. XMill: An Efficient Compressor for XML Data. In *Proceedings of the 2000 ACM SIGMOD International Conference on Management of Data*, pages 153–164, May 2000.

[5] P. G. Howard and J. S. Vitter. Analysis of Arithmetic Coding for Data Compression. In *Proceedings of the IEEE Data Compression Conference*, pages 3–12,April 1991.

[6] Tolani P.M. and Haritsa J.R., "XGRIND: a query-friendly XML compressor," in Proc. 2002 Int'l Conf. on DatabaseEng., pp. 225-34

[7] World Wide Web Consortium. Document Object Model (DOM) Level 1 Speci_cation Version 1.0, W3C Recommendation 1 October, 1998 edition.

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Congestion Control and Buffering Technique for Video Streaming over IP

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Abstract: **-** Due to the fiery growth of the demand for transmission of real-time video, streaming video over the Internet has received marvelous concentration. An efficient Quality of Service of video streaming depends on bandwidth, delay, and loss requirements due to its real-time nature. However, the current Internet system is not offering any quality of service (QoS) guarantees to streaming video over the Internet. Also it is difficult to support multicast video efficiently while providing service suppleness to meet a wide range of QoS requirements from the users. Thus, designing mechanisms and protocols for Internet streaming video poses many challenges. To converse to these challenges, extensive researches has been conducted and has found six key areas of streaming video for maintaining an acceptable QoS. We have introduced an improved application layer QoS control to avoid congestion and maximize video quality by reducing packet loss and delay. For this, we address the packet loss issue and review major approaches and mechanisms. On our approaches for application layer QoS improves the video streaming quality significantly. We also discuss the tradeoffs of the approaches and point out future research directions.

Key words:- Application Layer Congestion Control, Video Streaming, Receiver based Buffering technique, Video Compression, Streaming Server, Media Synchronization.

I. INTRODUCTION

The demand for multimedia information on the web is increasing day by day due to various multimedia applications such as distance learning, digital libraries, home shopping, and video-on-demand. Recent advances in computing technology, compression technology, high-bandwidth storage devices, and high-speed networks have made it feasible to provide real-time multimedia services over the Internet. Real-time multimedia [1] as the name implies, has timing constraints. For example, audio and video data must be played out continuously. If the data does not arrive on time, the play out process will pause, which is irritating to human hearing and visuals. So this real-time transport of live video or stored video is the predominant part of real-time multimedia.

In this research paper, we are fretful with the video streaming, which refers to real-time transmission of stored video. Streaming video is a sequence of "moving image" that are sent in compressed form over the Internet and displayed by the viewer as they arrive. For the transmission of stored video over the Internet there are two modes such as the download mode and the streaming mode (i.e., video streaming). In the download mode [2] a user downloads the entire video file and then plays back the video file. However, full file transfer in the download mode usually suffers long and perhaps unacceptable transfer time. In compare [2] in the streaming mode, the video content does not need be downloaded in full size, but is being played out while parts of the content are being received and decoded. Due to its real-time nature, video streaming typically has bandwidth, delay and loss requirements. However, as we know the current traditional Best effort Internet service does not offer any quality of service (QoS) which guarantees to streaming video over the Internet. In adding up, it is difficult to support multicast video efficiently while providing service flexibility to meet a wide range of QoS requirements from the users. Thus, designing mechanisms and protocols for Internet streaming video poses many challenges.

To address these challenges, extensive researches has been conducted and have introduced six key areas of streaming video, such as: (i) video compression, (ii) application-layer QoS control, (iii) continuous media distribution services, (iv) streaming servers, (v) media synchronization mechanisms, and (vi) protocols for streaming media. Every one of these six areas is a basic structure slab, with which architecture for streaming video can be built. Figure 1 shows architecture for a video streaming.

A. Video compression

Firstly any raw video and audio data are pre-compressed by video compression and audio compression algorithms and then it is saved in the storage devices.

B. Application-layer QoS control

To cope with varying network conditions [3] and different presentation quality requested by the users, various application-layer QoS control techniques have been proposed. The application-layer techniques include congestion control and error control. Their respective functions are as follows. Congestion control is employed to prevent packet loss and reduce delay. Error control, on the other hand, is to improve

video presentation quality in the presence of packet loss. Error control mechanisms include forward error correction (FEC), retransmission, error-resilient encoding, and error concealment. Any streaming server [3] first retrieves the compressed video or audio data from the storage devices and then the application layer QoS control module adapts the video/audio bit-streams according to the network status and QoS necessities. After the adaptation, the compressed bit-streams are being packetize by the the transport protocols and then send the video/audio packets to the Internet. Packets may be dropped or may experience too much delay inside the Internet due to congestion.

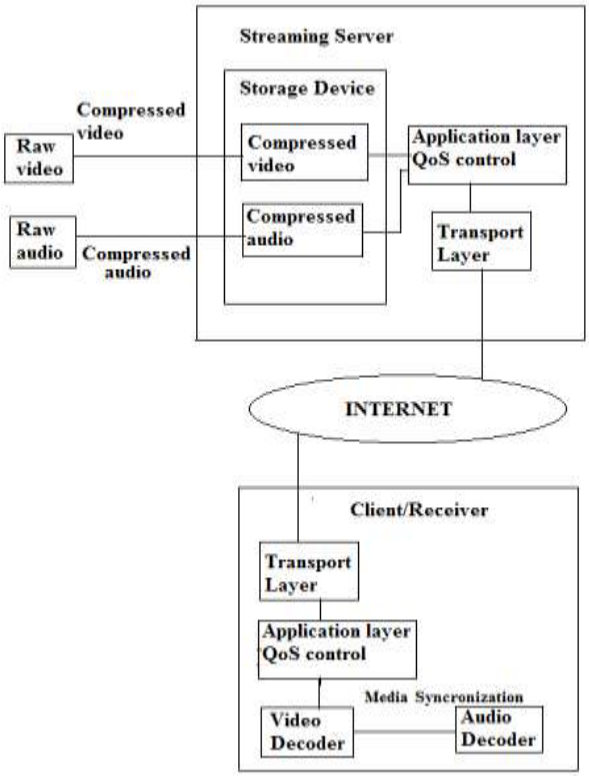

Figure 1: Architecture for video streaming.

C. Continuous media distribution services

In order to provide quality multimedia presentations, adequate network support is crucial. [4] This is because network support can reduce transport delay and packet loss ratio. Built on top of the Internet (IP protocol), continuous media distribution services are able to achieve QoS and efficiency for streaming video/audio over the best-effort Internet. Continuous media distribution services include network filtering, application-level multicast, and content replication.

D. Streaming servers

Streaming servers play an important role in providing streaming services. To offer quality streaming services, streaming servers need to process multimedia data under timing constraints and support interactive control operations such as pause/resume, fast forward, and fast backward. Furthermore, streaming servers need to retrieve media components in a synchronous fashion. A streaming server typically consists of three subsystems, namely, a communicator (e.g., transport protocols), an operating system, and a storage system.

E. Media synchronization mechanisms

Media synchronization is a major feature that distinguishes multimedia applications from other traditional data applications. With media synchronization mechanisms, the application at the receiver side can present various media streams in the same way as they were originally captured. An example of media synchronization is that the movements of a speaker's lips match the played-out audio.

F. Protocols for streaming media

Protocols are designed and standardized for communication between clients and streaming servers. Protocols for streaming media provide such services as network addressing, transport, and session control.

In the following Sections, we have discussed different types of congestion control mechanism which are based on the Application-layer QoS control in video streaming architecture and have tried to propose an efficient way to reduce packet loss and delay in video streaming.

II. APPLICATION-LAYER QOS - CONGESTION CONTROL

Quality of Service is a major challenge in video streaming. Packet loss and delay of arrival of video packet are the two key threats [4] for any kind of Efficient QoS in video streaming. Burst loss and excessive delay have a devastating effect on video presentation quality, and they are usually caused by network congestion. Thus, congestion-control mechanisms at end systems are necessary to help reducing packet loss and delay.

We have found that Application-layer QoS control deals with the packet loss and delay of video packet. The objective of application-layer QoS control is to avoid congestion and maximize the video quality by reducing packet loss and delay. The application-layer QoS control techniques include congestion control and error control. These techniques are employed by the end systems and do not require any QoS support from the network.

We have surveyed on various approaches for congestion control and propose a mechanism for packet loss and delay reduction.

A. Congestion Control

Congestion control depends on efficient rate control and rate shaping. Rate shaping is required for Source-Based Rate Control. Typically, for streaming video, congestion control takes the form of rate control. Rate control attempts to minimize the possibility of network congestion by matching

the rate of the video stream to the available network bandwidth.

In the following subsection we will deals with the two main factors of congestion control.

1. Rate Control

Rate control is a technique used to determine the sending rate of video traffic based on the estimated available bandwidth in the network. Existing rate-control schemes can be classified into three categories: source-based, receiver-based, and hybrid rate control.

Source-Based Rate Control

Under the source-based rate control, the sender is responsible for adapting the video transmission rate. Typically, feedback is employed by source-based rate-control mechanisms. Based upon the feedback information about the network [5] the sender could regulate the rate of the video stream. The source-based rate control can be applied to both unicast and multicast.

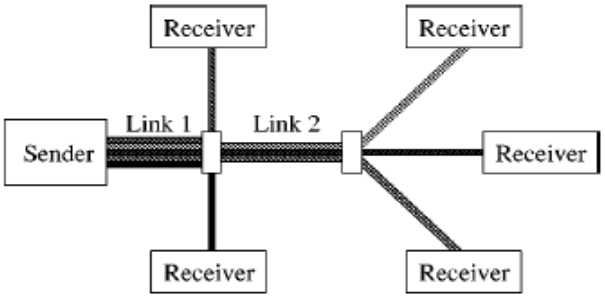

Figure 2: Unicast video distribution using multiple point-to-point connections.

For unicast video, existing source-based rate-control mechanisms follow two approaches: probe-based and model-based approach.

The probe-based approach is based on probing experiments. Specifically, the source probes for the available network bandwidth by adjusting the sending rate in a way that could maintain the packet loss ratio below a certain threshold.

The model-based approach is based on a throughput model of a transmission control protocol (TCP) connection. Specifically, the throughput of a TCP connection can be characterized by the following equation [5]:

 Eq.1

Where,

λ= throughput of a TCP connection

MTU (maximum transit unit) is the packet size used by the Connection

RTT round-trip time for the connection

ρ=packet loss ratio experienced by the connection.

Under the model-based rate control, the equation 1 is used to determine the sending rate of the video stream.

For multicast under the source-based rate control, the sender uses a single channel to transport video to the receivers which is given in Figure 3. Such multicast is called "single-channel multicast". For single-channel multicast, only the probe-based rate control can be employed.

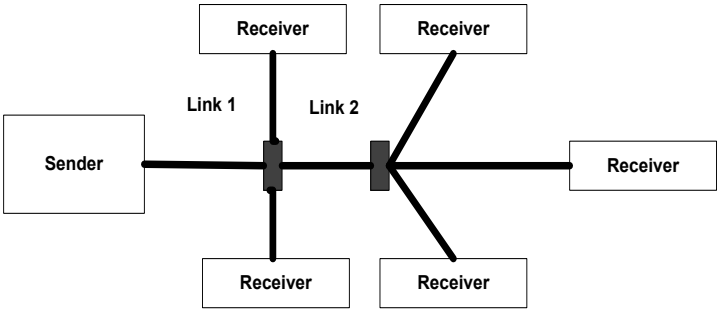

Figure 3: Multicast video distribution using multiple point-to-multipoint connections.

Receiver -Based Rate Control

Under the receiver-based rate control, the receivers regulate the receiving rate of video streams by adding/dropping channels while the sender does not participate in rate control. Typically, receiver-based rate control is used in multicasting scalable video, where there are several layers in the scalable video and each layer corresponds to one channel in the multicast tree. Similar to the source-based rate control, the existing receiver based rate-control mechanisms follow two approaches: probe based and model-based approach.

The basic probe-based rate control consists of two parts [6]

1) When no congestion is detected, a receiver probes for the available bandwidth by joining a layer/channel, resulting in an increase of its receiving rate. If no congestion is detected after the joining, the join-experiment is successful. Otherwise, the receiver drops the newly added layer.

2) When congestion is detected, a receiver drops a layer (i.e., leaves a channel), resulting in a reduction of its receiving rate.

Unlike the probe-based approach, which implicitly estimates the available network bandwidth through probing experiments, the model-based approach uses explicit estimation for the available network bandwidth. The model-based approach is also based on equation 1.

Hybrid -Based Rate Control

Under the hybrid rate-control, the receivers regulate the receiving rate of video streams by adding/dropping channels, while the sender also adjusts the transmission rate of each channel based on feedback from the receivers. Examples of hybrid rate control include the destination set grouping and a layered multicast scheme.

2. Rate Shaping

The objective of rate shaping is to match the rate of a pre-compressed video bit stream to the target rate constraint. A rate shaper (or filter), which performs rate shaping, is required for the source-based rate control which is given in Figure 4.

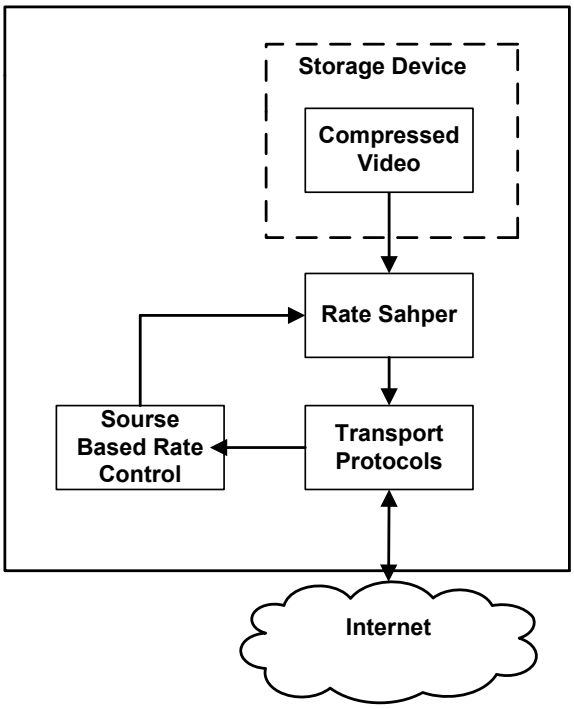

Figure 4: Architecture for source-based rate control

This is because the stored video may be pre-compressed at a certain rate, which may not match the available bandwidth in the network.

III. EFFICIENT CONGESTION CONTROL AND BUFFERING TECHNIQUE FOR REDUCING PACKET LOSS AND DELAY IN VIDEO STREAMING OVER THE INTERNET PROTOCOL

Efficient Congestion Control and Buffering technique for reducing packet loss and delay in Video Streaming over the Internet Protocol.

A. Efficient Congestion Control

Congestion control technique may vary due to the type of receiver. Single-channel multicast is efficient since all the receivers share one channel. However, single-channel multicast is unable to provide flexible services to meet the different demands from receivers with various access link bandwidths. In contrast, if multicast video were to be delivered through individual unicast streams, the bandwidth efficiency is low, but the services could be differentiated since each receiver can negotiate the parameters of the services with the source. Unicast and single-channel multicast are two extreme cases shown in Figure 5.

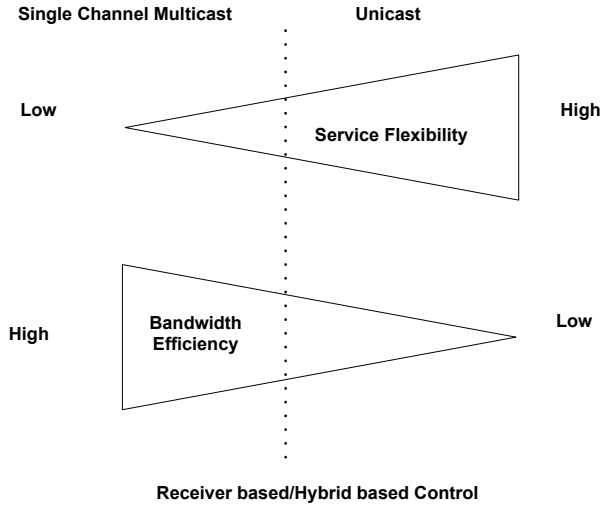

Figure 5: Tradeoff between efficiency and flexibility.

To achieve good tradeoff between bandwidth efficiency and service flexibility for multicast video, receiver-based and hybrid rate-control are proposed. One the other hand for efficiency in unicast video, source-based control gives good performance.

B. Buffering video stream before playing to receiver

In video streaming, [7] the video packet is sent in a continuous stream and is played as it arrives. We have found that delay is common problem for the video streaming. Sometimes it is found that ith packet may reach at the receiver after i+1th packet. As in real time transmission, the packet plays as soon as it reaches so here ith packet plays after i+1th packet. So it breaks the continuity of video streaming and degrades the quality of video.

So we have proposed a buffering concept for storing video packets at the receiver before playing the video. We have proposed to store last three packets in the buffer before playing these. According to our proposal the packet is not played as soon as it reached at the receiver end. At the receiver end the packet is stored in the buffer and the buffer can store three packets. When the buffer is full then the packets will be played.

We have found that this concept of using buffer at the receiver end before playing will reduce the probably of playing i+1th packet before ith packet. We have found that the buffering system improves the continuity of the video streaming which provides efficient quality of service.

IV. RESULT AND DISCUSSION

We have got the result of using Congestion Control technique and buffering for storing video stream. We have got the result from subjective test which is known as Mean Opinion Score.

A. Subjective Benchmark MOS (mean Opinion Score)

Figure 6 shows a common subjective benchmark MOS (Mean Opinion Score) obtained from a group of receiver for finding

the performance of video quality in case of our proposed efficient congestion control and buffering technique for reducing packet loss and delay in Video Streaming over the Internet Protocol.

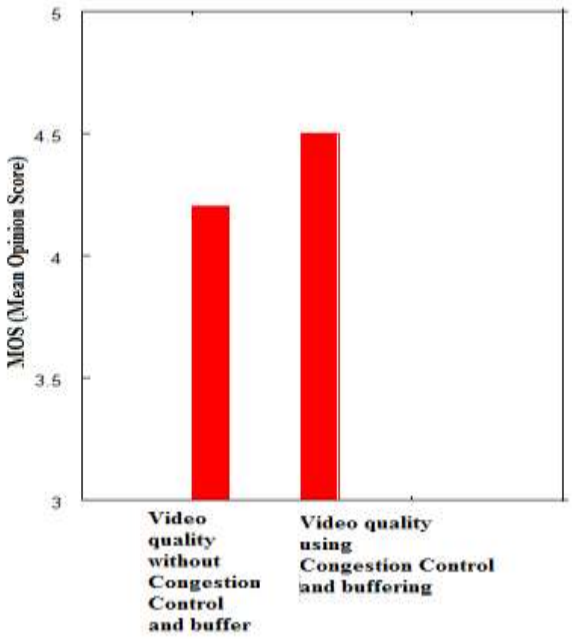

Figure 6: MOS (Mean Opinion Score) for efficient congestion control and buffering technique for reducing packet loss and delay in Video Streaming over the Internet Protocol.

VI. CONCLUSION

Video streaming is a vital constituent of many Internet multimedia applications. The best-effort nature of the current Internet poses many challenges to the design of streaming video systems. In this paper, we have surveyed major approaches and mechanisms for Internet video streaming. We would like to stress that the six areas are basic building blocks for a streaming video architecture. We have found that Application-layer QoS control deals the packet loss and delay by Congestion Control. We have discussed various Congestion Control procedures and found out an efficient Congestion Control mechanism. In our paper we have also proposed a buffering procedure to store the received voice stream temporary before playing. We have found that our proposed mechanism reduced packet loss and delay to improve quality of service of voice streaming. In future we will try to find an efficient error control technique for voice streaming which is another quality of service issue in voice streaming.

REFERENCES

[1]Dapeng Wu, Yiwei Thomas Hou, Wenwu Zhu, Ya-Qin Zhang, Jon M. Peha :Streaming Video over the Internet:Approaches and Directions. *IEEE TRANSACTIONS ON CIRCUITS AND SYSTEMS FOR VIDEO TECHNOLOGY*, VOL. 11, NO. 3, MARCH 2001

[2]A. Albanese, J. Blömer, J. Edmonds, M. Luby, and M. Sudan :Priority encoding transmission. *IEEE Trans. Inform. Theory*, vol. 42, pp. 1737–1744, Nov. 1996.

[3] S. Berson, L. Golubchik, and R. R. Muntz :Fault tolerant design of multimedia servers. *Proc. ACM SIGMOD'95*, May 1995, pp. 364–375.

[4] G. Blakowski and R. Steinmetz: A media synchronization survey: Reference model, specification, and case studies. *IEEE J. Select. Areas Commun*, vol. 14, pp. 5–35, Jan. 1996.

[5] S. Y. Cheung, M. Ammar, and X. Li :On the use of destination set grouping to improve fairness in multicast video distribution. *Proc. IEEE INFOCOM'96*, pp. 553–560, Mar. 1996.

[6] M. Chen, D. D. Kandlur, and P. S. Yu :Support for fully interactive playout in a disk-array-based video server", in Proc. ACM Multimedia, 4, New York, Oct. 1994.

[7] G. J. Conklin, G. S. Greenbaum, K. O. Lillevold, A. F. Lippman, and Y. A. Reznik :Video coding for streaming media delivery on the Internet. *IEEE Trans. Circuits Syst. Video Technol*., vol. 11, Mar. 2001.

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The Empirical Study on the Factors Affecting Data Warehousing Success

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Abstract: - Data Warehouse is the centralized store of detailed data from all relevant source systems, allowing for ad hoc discovery and drill-down analysis by multiple user groups. Various implementation factors play critical role to successful data warehouse (DW) project implementation. DW has unique characteristics that need to consider during implementation. There is little empirical research about implementation of DW to get success. Determining factors affecting DW success are important in the deployment of this DSS technology by organizations.

Keywords: - Data Warehouse, Business intelligence, Decision support system

1. Introduction

Data Warehouse is a technique for properly storing and managing data from different data sources for the purpose of business performance analysis.

Decision support system (DSS) is an area of the information systems (IS) discipline that focuses on supporting and improving managerial decision making [1]. In terms of contemporary professional practice, DSS includes personal decision support systems (PDSS), group support systems (GSS), executive information systems (EIS), online analytical processing systems (OLAP), data warehousing (DW), and business intelligence (BI). Over the three decades of its history, DSS has moved from a radical movement that has changed the way IS were perceived in business, to a mainstream commercial IT movement, in which all organizations engage [1].

Successfully supporting managerial decision making has become critically dependent upon the availability of integrated and high quality information organized and presented to managers in a timely and easily understood manner. DWs have emerged to meet this need. Surrounded by analytical tools and models, DWs have the potential to transform operational data into BI by effectively identifying problems and opportunities.

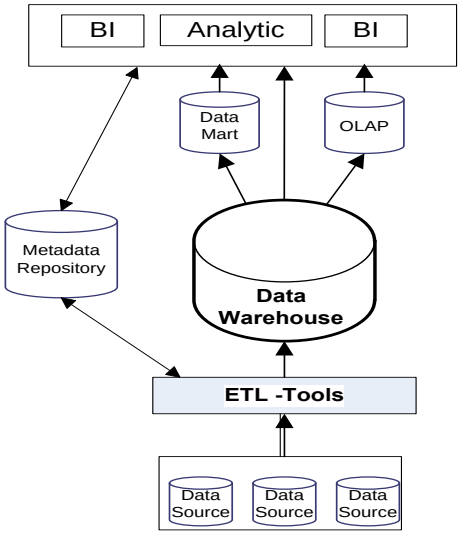

Figure1: A typical data warehousing system architecture

2. Aims of this Research

Large organizations have different data source to manage operation, so faced significant problem to build single view of their business from different data sources. During the mid-to-late 1990s, "DW became one of the most important developments in the information systems field" [1]. It is estimated that 95 percent of the Fortune 1000 companies either have a DW in place or are planning to develop one [2]." In 2002, the Palo Alto Management Group predicted that the DW market would grow to a $113.5 billion market, including the sales of systems, software, services, and in-house expenditures" [2].

About 3,000 data warehousing projects are undertaken each year and if the lowest perceived data warehouse failure rate (70%) is accurate, then each year there are 2,100 failures [3]. DW project is an expensive and risky undertaking [4]. The typical project costs over $1 million in the first year alone and it is estimated that one-half to two-thirds of all initial DW efforts fail [5]. There is a common perception that the failure rate of data warehousing projects is 70 to 80 percent (Inmon

2001) and one study reported a 90 percent failure rate (Conning 2000) [3]. "According to a 2003 Gartner report, more than 50 percent of data warehouse projects failed, in a 2007 study, Gartner predicted once more that 50 percent of data warehouse projects would have limited acceptance or be outright failures as a result of lack of attention to data quality issues" [6].
According to [7], the average time for the construction of a data warehouse is 12 to 36 months and the average cost for its implementation is between $1 million to $1.5 million.
This study was undertaken to perform a detailed analysis of these cases to determine whether the presence (or absence) of any specific factors or combination of factors might be correlated with instances of failures.

3. Factors affect DW success

A very good discussion on the problems of data warehousing projects is found in [7]. The paper mentions the logical fact that nobody really speaks about data warehousing failures and goes on to group the reasons for the failure of a data warehousing project into four categories, namely design, technical, procedural and socio-technical factors.

Serial #	Factors affect the DW success
01	IT initiate DW project
02	Poor Data Quality
03	Metadata Management
04	Less important pay on business value
05	Database Schema flaws
06	Quality of feeder system
07	POC on vendor talk about their product
08	Project Is Over Budget
09	Slipped Schedule
10	Functions and capabilities not implemented
11	Incomplete user's requirement
12	Unacceptable Performance
13	Poor Availability
14	Inability to expand
15	Poor Quality Reports
16	Tools is not user friendly
17	Project Not Cost Justified
18	Management does not recognize the benefits
19	Inadequate or no user involvement
20	Gap between researchers and practitioners

3.1 IT initiate DW project

In some organization IT used to provide different report to business to understand the business performance. Due to some reason (data volume increase, improper SQL write, Improper Application design, System over utilized, etc) business sometimes does not received the report right time. After that IT started talked about DW to give report to business on right time without proper analysis. Business agreed because they need report on time to analysis. The DW project is very costly and business talked about because of these report delay this type of costly project they will not finance. IT says to business you can do business analysis if we have DW, but business says we want report as business people no aware about business analysis with DW.
The IT Initiated DW/BI project may pay more attention on technology rather than business. If get sucked into the technology, then missing the whole point. Developed technically great DW/BI system but less importance on business, this DW project will be treating as fail. Need to start with business value.

3.2 Data Quality issue

Quality of data produces quality of information. Poor quality of data badly affects all company to run smoothly. Often the cause of business problem such as faulty analysis, operational inefficiency and dissatisfied customer are because of inaccurate, inconsistent, incomplete data. The operation cost increase due to poor quality data. The poor quality of data creates the problem data integration.
"Many enterprises fail to recognize that they have an issue with data quality. They focus only on identifying, extracting and loading data to the data warehouse, but do not take the time to assess quality, said Ted Friedman, principal analyst at Gartner" [6]. "Consistency and accuracy of data is critical to success with BI, and data quality must be viewed as a business issue and responsibility, not just an IT problem."[6]. "New federal regulations and corporate governance requirements have greatly increased the pressure for improved data quality. Enterprises must eliminate multiple data silos, assign stewardship to critical data, and implement a process for continuous monitoring and measurement of data quality."[6].
According to Gartner Inc ,through 2007 more than 50 percent DW project will be outright failed because lack of attention to data quality issue[6].

3.3 Metadata Management

The term Metadata is defined as "data about data"[8]. Metadata help a person to locate and understand data. Metadata is often generally described as "information about data." More precisely, metadata is the description of the data itself, its purpose, how it is used, and the systems used to manage it. Metadata play important role in DW development. Not only does it shape the data integration process but it also enables the business users to locate, understand and use the data once it is loaded into the data warehouse. Metadata is very valuable to the business because it facilitates the understanding of data. Without this understanding the data would be useless.
Metadata need to integrate from different source and developed metadata repository. "At the conceptual level the structure of the repository is described by a metamodel or informational model. To develop metamodel at least 4 modeling level are required. To start with level, on the lowest level, level 0 there are actual data item (e,g, the customer data). The level above contains the metadata information: level 1 contains metadata (e.g. database schema), level 2 specifies the schema used to store the metadata (the so called metamodel, information model or metadata schema)."[11]

3.4 Less important pay on business value

DW developed for analysis the business performance. So need to pay significant attention on business requirement and developed DW. Get the requirement from the business people some time it is difficult task, because the entire business person on the project may not from MBA background. Without work closely with business people can't deliver business. DW teams need to pay more important on business value. Keep in mind, Technology is important; business value is mandatory.

3.5 Database Schema flaws

The data stored in database and application access data. The schema holds the data, business rules inside the database. In one database may be more than one Schema. "The database schema should be flawless in terms of consistency, integrity and compliance to the rules of the relational technology before a data warehouse project is initiated."[11]

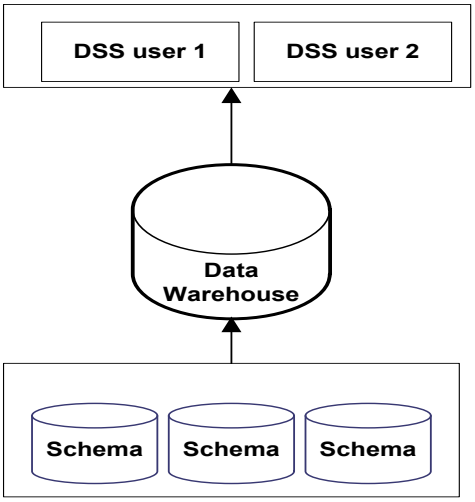

Figure 2: Database quality information

"The quality of the information depends on 3 things: (1) the quality of the data itself, (2) the quality of the application programs and (3) the quality of the database schema."[11]

3.6 Quality of feeder system

Data warehouse is depending on feeder system. If the feeder system has flaws then information provided by the DW will be flaws. All the feeder system integrated and stored required data in DW. During running ETL process the feeder system should capable enough to handle the process, in terms of performance and availability. Another point is quality of data of feeder system. If the quality of data is poor, need to clean before load in DW.

3.7 POC (proofs of concept) on vendor talk about their product

The proof of concept will help us to ensure vendor claim is true or not. The popularity of DW grew exponential during last year [10]. So there is huge number of product is now in market on DW. Vendor say about a lot about their product they offered. Organization need to carefully consider the vendors marketing claim regarding the product feature. Vendor can provide the customer reference of the product. Carefully need to choose the product otherwise during implementation project will be treated as fail. It is better to test the vendor claim on organization before chose the product to build DW.

3.8 Project is over budget

Every company used to declare annual budget, if the project required more budget, the project may be treated a failure. "The DW is costly project so the inadequate budget might be the result of not wanting to tell management the bitter truth about the costs of a data warehouse and expensive consulting help may have been needed. Performance or capacity problems, more users, more queries or more complex queries may have required more hardware to resolve the problems. The scope of project may require updating on the middle of the project running or other factors may have resulted in additional expenses."[12]

3.9 Slipped Schedule

"Most of the factors listed in the other section could also have contributed to the schedule not being met, but the major reason for a slipped schedule is the inexperience or optimism of those creating the project plan. In many cases management wanting to "put a stake in the ground" were the ones who set the schedule by choosing an arbitrary date for delivery in the hope of giving project managers something to shoot for. The schedule becomes a deadline without any real reason for a fixed delivery date. In those cases the schedule is usually established without input from those who know how long it takes to actually perform the data warehouse tasks. The deadline is usually set without the benefit of a project plan. Without a project plan that details the tasks, dependencies and resources, it is impossible to develop a realistic date by which the project should be completed."[12]

3.10 Defined functions and capabilities not implemented

Before project implementation, the scope of project or capabilities and functionalities of project need to well defined. This scope is defined from the input given from business and technical concern person. In this scope are specified certain functionality and capabilities. If important function and capabilities are not considered or postponed to some other implementation phase, this can be indication of project failure.

3.11 Incomplete user's requirement

Before project start DW user compile the requirement. User expect that they will get their requirement fulfill. If the users not get their requirement form get unhappy and due to this the project should be considered a failure. Sometimes users expect more than they got. "Users may be unhappy about the cleanliness of their data, response time, availability, usability

of the system, anticipated function and capability, or the quality and availability of support and training."[12]

3.12 Unacceptable Performance

"Unacceptable performance has often been the reason that data warehouse projects are cancelled. Data warehouse performance should be explored for both the query response time and the extract/transform/load time. Query response time is take couple of minutes is acceptable, because some time millions of rows need to calculate. We have seen queries where response time is measured in days. Except for a few exceptions, this is clearly unacceptable. This will impact the availability of the data warehouse to the users. Not only user response but also minimize the execution time of ETL process. "[12]

3.13 Poor Availability

"Availability is both scheduled availability (the days per week and the number of hours per day) as well as the percentage of time the system is accessible during scheduled hours. Availability failure is usually the result of the data warehouse being treated as a second-class system. Operational systems usually demand availability service level agreements. The performance evaluations and bonus plans of those IT members who work in operations and in systems often depends on reaching high availability percentages. If the same standards are not applied to the data warehouse, problems will go unnoticed and response to problems will be casual, untimely and ineffective."[12]

3.14 Inability to Expand

"If a robust architecture and design is not part of the data warehouse implementation, any significant increase in the number of users or increase in the number of queries or complexity of queries may exceed the capabilities of the system. If the data warehouse is successful, there will also be a demand for more data, for more detailed data and, perhaps, a demand for more historical data to perform extended trend analysis, e.g. five years of monthly data."[12]

3.15 Poor Quality Reports

"If the data is not clean and accurate, the queries and reports will be wrong, In which case users will either make the wrong decisions or, if they recognize that the data is wrong, will mistrust the reports and not act on them. Users may spend significant time validating the report figures, which in turn will impact their productivity. This impact on productivity puts the value of the data warehouse in question."[12]

3.16 Tools is not user friendly

Should not expect that all the user who will use BI tools for business analyses are IT expert. Some of the user may heard about the BI tools is first time. If the tools is not user friendly people will not used much and start blame on this tools even though form technical and business point of view tools is great. User may start asking ask IT to download the report and provide them. The DW is not only some sort of report, it is more that that where user will build up their own query and design report to analysis the business performance. In this scenario the project will be treating as failed, because the purpose of DW is deviated.

3.17 Project Not Cost Justified

"Every organization should justify cost their data warehouse projects. Justification includes an evaluation of both the costs and the benefits. When the benefits were actually measured after implementation, they may have turned out to be much lower than expected, or the benefits came much later than anticipated. The actual costs may have been much higher than the estimated costs. In fact, the costs may have exceeded both the tangible and intangible benefits."[12]

3.18 Management does not recognize the benefits

Management should properly inform time to time about the benefit getting from DW. "The project managers may believe that everyone in the organization will automatically know how wonderfully IT performed, and that everyone will recognize the data warehouse for the success that it is. They are wrong. In most cases, if management is not properly briefed on the data warehouse, they will not recognize its benefits and will be reluctant to continue funding something they do not appreciate."[12]

3.19 Inadequate or no user involvement

"It is hard to believe that IT organizations still build data warehouses with little or no business involvement, said Frank Buytendijk, research vice president at Gartner" [8]. "But some IT experts still believe it is important to 'anticipate the needs of the users.' They also suffer from the 'Atlas Syndrome' - trying to carry the weight of the world on its shoulders- solving problems the users 'do not understand.' As valid as this may seem, it results in a negative outcome."[8]

3.20 Gap between researchers and practitioners

"The gap between researchers and practitioners is widely discussed in the IT community" [9]. The situation regarding data warehousing seems to follow the general pattern where practitioners complain that their practical problems are overlooked by research and researchers are general unsatisfied by the acceptance of their ideas in industry. Pvassil [9] show possible new areas of research, based on practical problems and at the same time to give an idea of how practice could benefit from research results which seem to be rather ignored.

4. Conclusions

There are many ways for a data warehouse project to fail. We have research and found some factor, theses are discussed above. Before start the DW project, we propose to organization to understand the factor those affect the DW project success. The working with DW project is different than other IT projects, this also discussed. By knowing the factor of failures, organization can try to avoid those factors and DW project will be successes

References

[1] Arnott, D. and Pervan, G. (2005). A Critical Analysis of Decision Support Systems Research. Journal of Information Technology, 20(2), pp. 67 – 85.

[2] Eckerson, W.W. (2003). Evolution of Data Warehousing: The Trend toward Analytical Applications. Journal of Data Warehousing, 25(1), pp. 1-8

[3] Keith Lindsey, Mark N. Frolick. CURRENT ISSUES IN DATA WAREHOUSING. 2002 ,Eighth Americas Conference on Information Systems. http://aisel.aisnet.org/amcis2002/7

[4] Grim, R., and Thorton, P. (2001). P. A Customer for Life: The warehouse Approach. Journal of Data Warehousing, 2(1), pp. 73-79.

[5] Watson, H. J., and Haley, B. J. (2004). Data Warehousing: A Framework and Survey of Current Practices, Journal of Data Warehousing,2(1), pp. 10 – 17.

[6] Analysts to Show How To Implement a Successful Business Intelligence Program During the Gartner Business Intelligence Summit, March 7-9 in Chicago, IL http://www.gartner.com/press_releases/asset_121817_11.html [5/28/2010 12:28:20 AM]

[7] M. Demarest. The politics of data warehousing. http://www.noumenal.com/marc/dwpoly.html[9/23/2010 5:49:34 PM]

[8] http://www.mpcer.nau.edu/metadata/index_md1.htm[9/21/2010 5:44:34 PM]

[9] Panos Vassiliadis. Gulliver in the land of data warehousing: practical experiences and observations of a researcher. DMDW'2000. Stockholm, Sweden, June 5-6, 2000

[10] W.H. Inmon. Building the data Warehouse. John Wiley & Sons, 1996

[11] Why Data Warehouse Projects Fail. http://etnaweb04.embarcadero.com/resources/technical_papers/Why-Data-Warehouse-Projects-Fail.pdf [9/22/2010 5:44:34 PM]

[12] Sid Adelman, Larissa Moss. Data Warehouse Failures. http://www.tdan.com/view-articles/4876[5/13/2010 2:46:53 PM]

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Should you stay safe with BI tools and only select those that are highest rated by Gartner

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Abstract: - Assessing a market and its participants is a daunting task. Vendor differentiation caused by differing sizes, levels of complexity and strategies can inhibit comparisons of vendor offerings, and the market's overall direction is often murky. Markets vary in many ways, but all follow a predictable life cycle with these phases: embryonic, emerging, high growth, consolidating, maturity and declining [1]. Information exists through the organization. So some solutions need to maximize the value of that information and provide some benefit to organization. Due to demand from enterprises which want to invest in Business Intelligence (BI) solutions, vendors invest to developed the Business Intelligence solution. Select the right BI tools from vast range of product, is not so easy. To select the right product from the market sometimes user want some want some reports explores the competitive dynamics within the BI market and helps businesses select a vendor based on its technology strength, reputation among customers, and impact in the market. Gartner provides a complete view of vendor capabilities and advises on those you should explore, consider and, most importantly, shortlist.

Keywords: - Data Warehouse, Business intelligence, Gartner

1. Introduction

In fact, according to a recent survey by Gartner, The business priority "improving business processes" has been the No. 1 business expectation of IT since its introduction to the CIO Agenda survey in 2005. In 2009, more than 57 percent of CIOs reported this as one of their top five business expectations [2]. That worldwide survey of more than 1,500 CIOs by Gartner Executive Programmes (EXP) identifies business intelligence applications as the number one technology priority [2]. Software vendor develop the BI solution and each claim to offer best solution. So there is huge number of product is now in market on BI solution. Vendor say about a lot about their product they offered. Organization need to carefully consider the vendors marketing claim regarding the product feature. Carefully need to choose the product. Magic Quadrants offer visual snapshots of a market's direction, maturity and participants, Understanding Gartner's research methodology will help organizations use these models effectively when choosing a product or service, or managing a vendor relationship[1].

2. Aims of this Research

"According to Datamonitor's Global IT Applications Model (IMTC0105), between 2007 and 2012 the BI market is set to expand by a compound average growth rate (CAGR) of 12% to 13%. This is significantly higher than the growth rate in most of the other enterprise application markets over the same period."[3]. Chose wrong product is also one important factor to fail the project. There are some companies given some indicator to choose right BI product. To read these documents and do analysis to find the best product is not so easy from BI market. There are some companies doing analysis to find the best product from BI market. Among them Gartner is one of them. Gartner have defined criteria to find the best product from the vast BI market. Our main objectives of this research is to find the answer of "Should you stay safe with BI tools and only select those that are highest rated by Gartner"

3. BI

Organization measures business performance on own mechanism. To run the business smooth every organization need to do some business performance analysis. Every IT enabled company used business intelligence (BI) software in some capacity and BI able to deliver high return on investment. Data store in different database or flat file across the organization. BI unlocks the valuable information assets from these sources of data. It incorporates performance management measures for tracking and acting on key performance indicators. BI also provides more financial transparency. "We'll have a complete picture of our finances, where donations are coming in and where money is going out," says Mike Sabot, business analyst at Habitat [4].

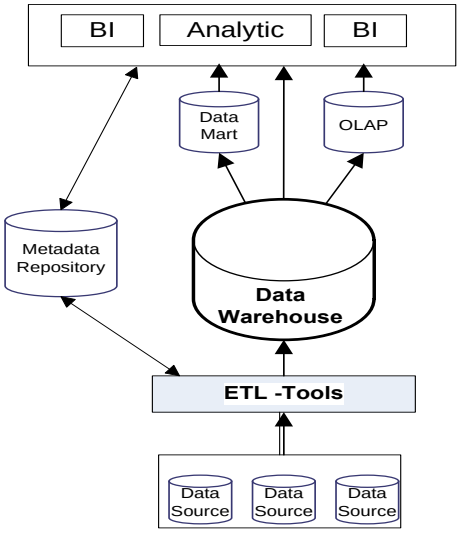

Figure1: A typical data warehouse and BI system architecture

4. Organizations: Without BI tools

Business intelligence tool provide technical feature that include query reporting, online analytic processing (OLAP), data miming and data visualization. To analysis the business performance executives need data. The vast majority of businesses have information scattered throughout the enterprise on paper, in siloed databases and in email [5]. Data cannot be easily extracted and mostly is unusable format. Most companies have four or five different intelligence products to meet their needs [5]. Companies have tended to buy BI products individually to solve a perceived pain point [y]. Many organization used excel or other spreadsheet to analysis the data. In a November 2003 survey, Softrax found 87% of *Global 1000* corporate financial executives use Excel or other spreadsheets to track and monitor compliance and processes not covered in their financial and accounting packages [5]. With over 150 million users, Excel is so pervasive Gartner calls it the "ubiquitous business process application in the enterprise."[5]. Spreadsheets are highly inefficient and prone to error. Moreover it is required huge amount of time compiling and processing information.

5. Organizations: Too Many BI Tools:

"More companies are signing deals to implement business intelligence software companywide, vendors claim. But only 33% of those queried in the InformationWeek survey have standardized on one or a few BI tools deployed throughout their companies, compared with 32% in last year's survey."[4] Most companies still have a variety of business intelligence tools from various vendors being used throughout departments, operations, and locations, or being deployed on a project-by-project basis.

6. Gartner

"Gartner, Inc. (NYSE: IT) is the world's leading information technology research and advisory company. Gartner deliver the technology-related insight necessary for our clients to make the right decisions, every day. No other research or consulting firm can offer insight that is as accurate, impartial, objective and consistent. Gartner does not sell technology, nor do implement technology. Independence is the key to Gartner's objectivity. From CIOs and senior IT leaders in corporations and government agencies, to business leaders in high-tech and telecom enterprises and professional services firms, to technology investors, Gartner's are the valuable partner to 60,000 clients in 10,800 distinct organizations. Through the resources of Gartner Research, Gartner Executive Programs, Gartner Consulting and Gartner Events, we work with every client to research, analyze and interpret the business of IT within the context of their individual role. Founded in 1979, Gartner is headquartered in Stamford, Connecticut, U.S.A., and has 4,300 associates, including 1,200 research analysts and consultants, and clients in 80 countries."[5]

7. Magic Quadrants

Magic Quadrants depict markets in the middle phases of their life cycle by using a two dimensional matrix that evaluates vendors based on their completeness of vision and ability to execute. The Magic Quadrant has 15 weighted criteria that plot vendors based on their relative strengths in the market. This model is well suited for high-growth and consolidating markets where market and vendor differentiations are distinct. Magic Quadrants and MarketScopes are updated at least annually and may be updated sooner to respond to market changes. Vendors positioned in the four quadrants — Leaders, Challengers, Visionaries and Niche Players — share certain characteristics.

Leaders

"Leaders provide mature offerings that meet market demand as well as demonstrate the vision necessary to sustain their market position as requirements evolve. The hallmark of leaders is that they focus and invest in their offerings to the point that they lead the market and can affect its overall direction. As a result, leaders can become the vendors to watch as you try to understand how new offerings might evolve.

Leaders typically possess a large, satisfied customer base (relative to the size of the market) and enjoy high visibility within the market. Their size and financial strength enable them to remain viable in a challenging economy.

Leaders typically respond to a wide market audience by supporting broad market requirements. However, they may fail to meet the specific needs of vertical markets or other more-specialized segments. "[1]

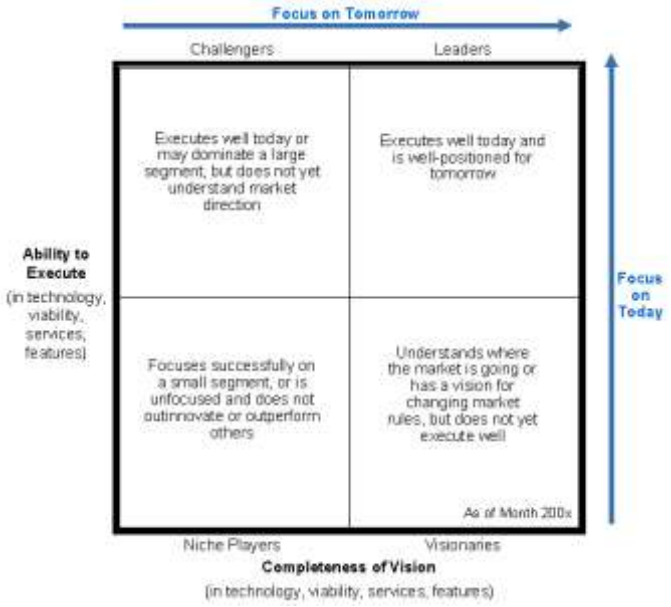

Figure 1. The Magic Quadrant

Challengers

"Challengers have a strong ability to execute but may not have a plan that will maintain a strong value proposition for new customers. Larger vendors in mature markets may often be positioned as challengers because they choose to minimize risk or avoid disrupting their customers or their own activities.

Although challengers typically have significant size and financial resources, they may lack a strong vision, innovation or overall understanding of market needs. In some cases, challengers may offer products nearing the end of their life that dominate a large but shrinking segment.

Challengers can become leaders if their vision develops. Large companies may fluctuate between the Challengers and Leaders quadrants as their product cycles and market needs shift."[1]

Visionaries

"Visionaries align with Gartner's view of how a market will evolve, but they have less-proven capabilities to deliver against that vision. In early markets, this status is normal. In more-mature markets, it may reflect a competitive strategy for a smaller vendor — such as selling an innovation ahead of mainstream demand — or a larger vendor trying to break out of a rut or differentiate.

For vendors and customers, visionaries fall in the higher-risk/higher-reward category. They often introduce new technology, services or business models, and they may need to build financial strength, service and support, and sales and distribution channels. Whether visionaries become challengers or leaders may depend on if companies accept the new technology or if the vendors can develop partnerships that complement their strengths. Visionaries sometimes are attractive acquisition targets for leaders or challengers."[1]

Niche Players

"Niche players do well in a segment of a market, or they have limited ability to innovate or outperform other vendors. This may be because they focus on a functionality or geographic region, or they are new entrants to the market. Alternatively, they may be struggling to remain relevant in a market that is moving away from them. Niche players may have reasonably broad functionality but with limited implementation and support capabilities, and relatively limited customer bases. They have not yet established a strong vision for their offerings.

Assessing niche players is more challenging than assessing vendors in other quadrants because some niche players could make progress, while others do not execute well and may not have the vision to keep pace with broader market demands.

A niche player may be a perfect fit for your requirements. However, if it goes against the direction of the market — even if you like what it offers — it may be a risky choice because its long-term viability will be threatened.[1]

8. Gartner's Vendor section and market analysis strategy

"MQ is cover vendor in a market based on the defined criteria. A Magic Quadrant is not intended to be an exhaustive analysis of every vendor in a market, but rather a focused analysis. Inclusion criteria consist of market share, revenue, number of clients, types of products or services, target market, or other defining characteristics that help narrow the scope of the research to those vendors that Gartner's consider to be the most important or best suited to Gartner clients' needs.

Research activities include but are not limited to:

1. Vendor briefings
2. Surveys
3. Vendor-provided references
4. Industry contacts
5. Client interviews
6. Public sources, such as U.S. Securities and Exchange Commission filings, articles, speeches and published papers
7. Input from Gartner analysts

Teams of analysts collaborate to evaluate and score each vendor using the weighted criteria. The resulting scores are used to generate a Magic Quadrant.

The Magic Quadrant or MarketScope is published as a research document that explains the vendor positions and ratings, as well as new developments in the market, and thus provides a

context in which to use the models. During this step, the research undergoes rigorous internal peer review and validation, followed by a factual review by the vendors included on the Magic Quadrant or MarketScope."[1]

9. Define the Rating Criteria

Magic Quadrants use standard criteria in two categories: completeness of vision and ability to execute. We then adapt the inclusion criteria to a market by prioritizing and weighting them based on a high, low or standard scale of importance. In some cases, a criterion may have a "no rating" weight because it has low relevance for the market.

9.1 Completeness of Vision

Summarizes factors such as the vendor's financial viability, market responsiveness, product development, sales channels and customer base.

Evaluation Criteria	Weighting
Market Understanding	High
Marketing Strategy	High
Sales Strategy:	Standard
Offering (Product) Strategy	High
Business Model:	No Rating
Vertical/Industry Strategy:	Standard
Innovation	High
Geographic Strategy	Standard

Table 1. Completeness of Vision Evaluation Criteria

Market Understanding

"The ability of a vendor to understand buyers' needs and translate these needs into products and services. A vendor that shows the highest degree of vision listens and understands buyers' wants and needs, which it can shape or enhance with its vision."[1]

Marketing Strategy

"A clear, differentiated set of messages consistently communicated throughout the organization and publicized through the Web site, advertising, customer programs and positioning statements. "[1]

Sales Strategy

"A strategy for selling products that uses the appropriate network of direct and indirect sales, marketing, service, and communication affiliates to extend the scope and depth of a vendor's market reach, skills, expertise, technologies, services and customer base."[1]

Offering (Product) Strategy

"A vendor's approach to product development and delivery that emphasizes differentiation, functions, methodology and feature set in relation to current and future requirements. "[1]

Business Mode

"The validity and logic of a vendor's underlying business proposition. Not that this criterion has been given no rating because all vendors in the market have a viable business model. "[1]

Vertical/Industry Strategy

"A vendor's strategy to direct resources, skills and offerings to meet the needs of market segments, including vertical industries. "[1]

Innovation

Marshaling of resources, expertise or capital for competitive advantage, investment, consolidation or defense against acquisition. "[1]

Geographic Strategy

"A vendor's strategy to direct resources, skills and offerings to meet the needs of regions outside of the vendor's "home" or native area, directly or through partners, channels and subsidiaries, as appropriate for that region and market. "[1]

9.2 Ability to Execute

"Summarizes factors such as the vendor's financial viability, market responsiveness, product development, sales channels and customer base. "[1]

Evaluation Criteria	Weighting
Product/Service	High
Overall Viability	High
Sales Execution/Pricing:	High
Market Responsiveness and Track Record	Standard
Marketing Execution	Standard
Customer Experience	High
Operations	Low

Table 2. Ability to Execute Evaluation Criteria

Product/Service

"Core goods and services offered by the vendor that compete in and serve the market. This category includes product and service capabilities, quality, feature sets and skills, offered natively or through original equipment manufacturers as defined in the market definition and detailed in sub criteria. "[1]

Overall Viability
"Includes an assessment of the vendor's overall financial health, the financial and practical success of the relevant business unit, and the likelihood of that business unit to continue to invest in and offer the product within the vendor's portfolio of products. "[1]

Sales Execution/Pricing
"The vendor's capabilities in pre-sales activities and the structure that supports them. This criterion includes deal management, pricing and negotiation, pre-sales support, and the overall effectiveness of the sales channel. "[1]

Market Responsiveness and Track Record
"Ability to respond, change direction, be flexible and achieve competitive success as opportunities develop, competitors act, customer needs evolve and market dynamics change. This criterion also considers the vendor's history of responsiveness. "[1]

Marketing Execution
"The clarity, quality, creativity and efficacy of programs designed to deliver the vendor's message to influence the market, promote its brand and business, increase awareness of its products, and establish a positive identification with the product, brand or vendor with buyers. This "mind share" can be driven by a combination of publicity, promotions, thought leadership, word of mouth and sales activities. "[1]

Customer Experience
"Relationships, products, and services and programs that enable clients to succeed with the products evaluated. This criterion includes the ways customers receive technical support or account support. It can also include ancillary tools, customer support programs (and their quality), availability of user groups and service-level agreements. "[1]

Operations
"The vendor's ability to meet its goals and commitments. Factors include the quality of the organizational structure, such as skills, experiences, programs, systems and other vehicles that enable the vendor to operate effectively and efficiently. "[1]

10. Inclusion and Exclusion Criteria

"Vendors were included in the Magic Quadrant if they met the following requirements:

- Generate $20 million (Year2007-2009), $15(year 2010) or more total software revenue* from BI platform software sales annually or, in the case of open-source BI platform software, generate $20 million total company revenue annually.
- Have customers that have deployed the vendor's BI platform as their enterprise BI solution and, in the case of vendors that also supply transactional applications, the BI platform is routinely used by organizations that do not use its transactional applications.
- Deliver at least eight out of 12 capabilities was year 2009 to 2007, on 2010 Deliver at least Nine out of 13 capabilities in the BI platform market definition."[1]

12. BI capabilities Define by Gartner

BI platforms are used to build applications that help organizations learn and understand their business. Gartner defines a BI platform as a software platform that delivers the some capabilities. The number of capabilities is changed over time. On 2010 report there are 13, 2009 to 2007[7-11] report, there were 12 and on 2006 report mentioned 20 capabilities. These capabilities are organized into three categories of functionality: integration, information delivery and analysis [7].

Categories of functionality	**Capabilities(2010**	**Capabilities(2009, 2008,2007)**
Integration	BI infrastructure	BI infrastructure
	Metadata management	Metadata management
	Development	Development
	Workflow and collaboration	Workflow and collaboration
Information Delivery	Reporting	Reporting
	Dashboards	Dashboards
	Ad hoc query	Ad hoc query
	Microsoft Office integration	Microsoft Office integration
	Search Based BI	
Analysis	OLAP	OLAP
	Advanced visualization	Advanced visualization
	Predictive modeling and data mining	Predictive modeling and data mining
	Scorecards	Scorecards

Table 3. BI capabilities defied by Gartner

BI infrastructure

"All tools in the platform should use the same security, metadata, administration, portal integration, object model and query engine, and should share the same look and feel."[7]

Metadata management

"This is arguably the most important of the 12 capabilities. Not only should all tools leverage the same metadata, but the offering should provide a robust way to search, capture, store, and reuse and publish metadata objects such as dimensions, hierarchies, measures, performance metrics and report layout objects."[7]

Development

"The BI platform should provide a set of programmatic development tools —coupled with a software developer's kit for creating BI applications — that can be integrated into a business process, and/or embedded in another application. The BI platform should also enable developers to build BI applications without coding by using wizard-like components for a graphical assembly process. The development environment should also support Web services in performing common tasks such as scheduling, delivering, administering and managing."[7]

Workflow and collaboration

"This capability enables BI users to share and discuss information via public folders and discussion threads. In addition, the BI application can assign and track events or tasks allotted to specific users, based on predefined business rules. Often, this capability is delivered by integrating with a separate portal or workflow tool."[7]

Reporting

"Reporting provides the ability to create formatted and interactive reports with highly scalable distribution and scheduling capabilities. In addition, BI platform vendors should handle a wide array of reporting styles (for example, financial, operational and performance dashboards)."[7]

Dashboards

"This subset of reporting includes the ability to publish graphically intuitive displays of information, including dials, gauges and traffic lights. These displays indicate the state of the performance metric, compared with a goal or target value. Increasingly, dashboards are used to disseminate real-time data from operational applications."[7]

Ad hoc query

"This capability, also known as self-service reporting, enables users to ask their own questions of the data, without relying on IT to create a report. In particular, the tools must have a robust semantic layer to allow users to navigate available data sources. In addition, these tools should offer query governance and auditing capabilities to ensure that queries perform well."[7]

Microsoft Office integration

"In some cases, BI platforms are used as a middle tier to manage, secure and execute BI tasks, but Microsoft Office (particularly Excel) acts as the BI client. In these cases, it is vital that the BI vendor provides integration with Microsoft Office, including support for document formats, formulas, data "refresh" and pivot tables. Advanced integration includes cell locking and write-back."[7]

Search-Based BI

"Applies a search index to both structured and unstructured data sources and maps them into a classification structure of dimensions and measures(often leveraging the BI semantic layer) that users can easily navigate and explore using a search (Google- like) interface"[7]

OLAP

"This enables end users to analyze data with extremely fast query and calculation performance, enabling a style of analysis known as "slicing and dicing." This capability could span a variety of storage architectures such as relational, multidimensional and in-memory."[7]

Advanced visualization

"This provides the ability to display numerous aspects of the data more efficiently by using interactive pictures and charts, instead of rows and columns. Over time, advanced visualization will go beyond just slicing and dicing data to include more process-driven BI projects, allowing all stakeholders to better understand the workflow through a visual representation."[7]

Predictive modeling and data mining

"This capability enables organizations to classify categorical variables and estimate continuous variables using advanced mathematical techniques."[7]

Scorecards

"These take the metrics displayed in a dashboard a step further by applying them to a strategy map that aligns key performance indicators to a strategic objective. Scorecard metrics should be linked to related reports and information in order to do further analysis. A scorecard implies the use of a performance management methodology such as Six Sigma or a balanced scorecard framework."[7]

13. Analysis on Gartner selection

Leader's quadrant is the best product, criteria defined by Gartner. The product exist on leader quadrant may not exit on next year. I have done analysis Gartner report "Gartner-Magic-Quadrant-for-Business-Intelligence-platforms" published from 2006 to 2010. Here I have seen some of the product exist continue on leader quadrant among the years but some are not. Only three products (Cognos, Information Builder, SAS) are continue on the leader quadrant.

However, looking for tools just as good that could end up being in the upper leader quadrant according to garner reports analysis [1]. From the analysis not a single product continue on end up being in the upper quadrant according to garner reports "Gartner-Magic-Quadrant-for-Business-Intelligence-platforms" published from 2006 to 2010[7-11].
Another point is BI product acquisition. Acquisition happened on leader quadrant's BI product from 2006 to 2010 three times.

Acquire product name	Gartner reported on year	Acquired by
Hyperion	2008	Oracle
Cognos	2009	IBM
Business Object	2009	SAP

Table 4. Acquire product

Business Objects was acquired by SAP, and its BI platform products are now sold alongside those developed by SAP itself [7]. Hyperion was acquired by Oracle [8]. IBM acquired Cognos[9]. Sometimes product is dropped after acquired by other company.
Now he/she chose the product this year from leader quadrant or end up being in the upper leader quadrant, may be during implementation phase of BI he/she see that product is not leader quadrant or not end up being in the upper leader quadrant or it is dropped by accrued vendor.

Also, it is interesting to see how Gartner arrive at the ratings. A major part of their ranking process relies on feedback from the existing customers. They ask the product vendors to nominate the customers from whom they seek the feedback.

Don't think for a minute that Gartner is not biased[12-16]].ZL claims that Gartner's use of their proprietary "Magic Quadrant" is misleading and favors large vendors with large sales and marketing budgets over smaller innovators such as ZL that have developed higher performing products[16].

Before use MQ, need to consider the comment: "Your needs and circumstances should determine how you use the Magic Quadrant, not the other way around. To evaluate vendors in the Leaders quadrant only and ignore those in other quadrants is risky and thus discouraged. For example, a vendor in the Niche Players quadrant could offer functions that are ideally suited to your needs. Similarly, a leader may not offer functions that meet your requirements — for example, its offerings may cost more than competitors', or it may not support your region or industry"[1]. This comment is just confusing. Leader quadrant product is best suit, criteria defined by Gartner. A lower Gartner rating does not mean inferior product.

12. Conclusion:

The magic quadrant is not very useful. Gartner's ratings are majorly based on customers' feedback on the products. Gartner reaches out to the list of customers supplied by product vendors (needless to say whom the vendors will chose) to carry out the survey. Naturally, there is an element of 'bias' that is introduced in the process. Microsoft jumped to a dramatic leader position in the last survey. All the product vendors naturally try to use this as a good marketing vehicle. In all, independent ratings like Gartner are still important and necessary to provide a "third party" view. I strongly suggest instead of looking at Gartner Quadrant for product ratings, one should look at the detailed analysis that Gartner provides for each of the vendors in the detailed report. There is wealth of information which will help you decide the right product for your organization. Gartner should be used only as guideline to select the product

References

[1] Magic Quadrants and MarketScopes: How Gartner Evaluates Vendors Within a Market. http://www.gartner.com/DisplayDocument?doc_cd=131166

[2] "Meeting the Challenge: The 2009 CIO Agenda," Gartner Executive Programmes (EXP), January 2006. http://www.gartner.com/it/page.jsp?id=855612[10/22/2010 9:42:45 PM]

[3] Decision Matrix: Selecting a Business Intelligence Vendor. http://www.sas.com/news/analysts/datamonitor_bi_0407.pdf

[4] BI Spending To Increase. InformationWeek. http://www.informationweek.com/news/global-cio/showArticle.jhtml?articleID=181500685[10/22/2010 9:56:15 PM]

[5] HOW TO SELECT BUSINESS INTELLIGENCE QUERY TOOLS. www.databi.com/downloads/bi**tools**.pdf [10/18/2010 9:45:31 PM]

[6] About Gartner. http://www.gartner.com [10/22/2010 9:45:31 PM]

[7] Magic Quadrant for Business Intelligence Platforms, 2010

[8] Magic Quadrant for Business Intelligence Platforms, 2009

[9] Magic Quadrant for Business Intelligence Platforms, 2008

[10] Magic Quadrant for Business Intelligence Platforms, 1Q06

[11] Magic Quadrant for Business Intelligence Platforms, 1Q07

[12] http://marklogic.blogspot.com/2009/10/gartner-sued-over-magic-quadrant-for.html[10/31/2010 10:07:25 AM]

[13] http://www.thebiggertruth.com/2009/12/zl-vs-gartner-interesting-at-a-many-levels/[10/16/2010 11:46:25 AM]

[14] http://sagecircle.wordpress.com/2009/10/21/this-not-the-first-time-that-gartner-has-been-sued/[10/16/2010 9:39:17 PM]

[15] http://www.scribd.com/doc/21362628/ZL-Opposition-to-Gartner-Motion-to-Dismiss[10/16/2010 9:39:17 PM]

[16] http://www.scribd.com/doc/23841976/ZLFirst-Amended-Complaint

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Medical Video Compression by Adaptive Particle Compression

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Abstract: This paper proposes Adaptive Particle Swarm Optimization (APSO) for medical video compression. The proposed technique is used to enhance quality of medical video with less computational complexity during compression. In APSO, the velocity and position equations of PSO are modified to achieve adaptive step size for getting true motion vector during video compression The new position of swarm depends on previous motion vector, time varying nonlinear inertia weight and time varying nonlinear acceleration coefficient which are also expressed in terms of motion vectors. Zero motion prejudgment is used that leads to faster convergence. The APSO is tested with three medical videos and results are compared with other algorithms. The proposed technique enhances quality of medical videos up to 1 db in terms of peak signal to noise ratio by saving computational time up to 85 % when compared with other published methods.

Keywords: Adaptive Particle Swarm Optimization (APSO). Video Compression, Peak Signal to Noise Ratio (*PSNR*), CT (Computerized Tomography), Computational Time, Medical Video

1. Introduction

Representing video material in a digital form requires a large number of bits. The volume of data generated by digitizing a video signal is too large for most storage and transmission systems. This means that compression is essential for most digital video applications. Statistical analysis of video signals indicates that there is a strong correlation both between successive picture frames and within the picture elements themselves. However, better compression performance may be achieved by exploiting the temporal redundancy in a video sequence or the similarities between successive video frames. This may be achieved by introducing two functions: 1. Prediction: create a prediction of the current frame based on one or more previously transmitted frames. 2. Compensation: subtract the prediction from the current frame to produce a residual frame. Then the residual frame is compressed by an image CODEC. In order to decode the frame the technology, decoder adds the prediction to the decoded residual frame. This is described as inter-frame coding for frames are coded based on some relationship with other video frames.

Video compression is necessary in a wide range of applications to reduce the total data required for the transmission or storage of video data. Video compression algorithm aims at exploiting the temporal and spatial redundancies by using some form of motion compensation followed by transform coding. The key step in removing temporal redundancy is the motion estimation where a motion vector is predicted between the current frame and a reference frame. Following the motion estimation, a motion compensation stage is used to obtain the original frame with the help of reference frame and motion vector [1]. Two major approaches are used for video sequence coding: block-based and object-based coding.

Several block-matching algorithms have been proposed. The exhaustive search (ES) or full search algorithm gives the highest peak signal to noise ratio amongst any block-matching algorithm but requires more computational time [1]. Over the past decade, many fast and efficient block-matching algorithms (BMA) have been proposed in order to achieve the accuracy and speed. Some of the well-known algorithms are Simple and Efficient Search (SES)[1], Three Step Search (TSS)[2], New Three Step Search (NTSS)[2], Four Step Search (4SS)[3], Diamond Search (DS)[4] and Adaptive Road Pattern Search (ARPS)[5]. These algorithms are widely accepted by the video compression community and have been used in implementing various standards, ranging from MPEG 1/H.261 to MPEG 4/H.263.

In 1994, Tsai *et al.* [6] developed a compression scheme for angiogram video sequence based on a full frame discrete wavelet transform. Gibson *et al.* [7] proposed a lossy wavelet-based approach for the compression of digital angiogram videos. In [8], a hybrid model has been discussed for the compression of CT sequences.

Real videos contain mixture of motions with slow and fast contents. Larger motions require a larger search parameter but it makes the process of motion estimation more computationally expensive. Massive computation is required for the implementation of ES. Many fast algorithms have been developed like the TSS, NTSS, 4SS, DS, etc. to reduce computational complexity. These algorithms are faster because only selected possible displaced blocks are matched within the search area in the reference frame, in order to find the block with minimum distortion. It is well known that in terms of computation these algorithms are better but the PSNR of these algorithms are low as compared to ES and are also motion dependent. The fixed step size algorithms are suitable for small or large motion depending upon the step size. Similarly, the prediction of true motion vector depends upon the step size. Therefore, in a video compression technique, step size plays vital role for getting true motion vectors without loosing the quality of video. Compared with fixed step-size motion estimation, the adaptive step algorithm improves video compression efficiency and hence overall video encoding speed.

Therefore, this paper presents the adaptive PSO (APSO) for medical video compression. The novelty of the proposed algorithm lies in adaptive step which dynamically

changes for small and large motion to locate the true motion vector and to reduce computational complexity without compromising the quality of video in terms of PSNR. Section 2 reviews the basic PSO technique. The proposed technique is discussed in section 3. In section 4, the APSO used for motion estimation is presented. Section 5 provides experimental results comparing APSO with other methods for video compression. Finally, conclusion is presented in section 6.

2. Particle Swarm Optimization

The Particle Swarm Optimizer (PSO) is a population-based optimization method [9-11] developed by Eberhart and Kennedy in 1995. PSO is inspired by social behavior of bird flocking or fish schooling, can handle efficiently arbitrary optimization problems. In Particle Swam Optimization, a particle is defined as a moving point in hyperspace. It follows the optimization process by means of local best (*Lbest*), global best (*Gbest*), particle displacement or position and particle velocity. In a PSO, particle changes their positions by flying around in a multi-dimensional search space until computational limitations are exceeded. The two updating fundamental equations in a PSO are velocity and position equations [9-11]. The particle velocity is expressed as Eq (1).

$$V_{K+1}^{i} = W * V_{k}^{i} + C_1 r_1 (Lbest - S_k^i) + C_2 r_2 (Gbest - S_k^i) \quad (1)$$

and the particle position is expressed as Eq. (2)

$$S_{k+1}^{i} = S_k^i + V_{k+1}^i \quad (2)$$

Where, *V*= Particle velocity
S= Particle Position
Lbest = Local best
Gbest = Global best
W = Inertia weight
C_1 and C_2 are acceleration constant
r_1 and r_2 are random values [0 1]
k = Current iteration
i = Particle number

In first parts, *W* plays the role of balancing the global search and local search. Second and third parts contribute to the change of the velocity. The second part of Eq. (1) is the "cognition" part, which represents the personal thinking of the particle itself. The third part of Eq (1) is "social part", which represents the collaboration among the particles [9-11]. Without the first part of Eq (1), all the particles will tend to move toward the same position. By adding the first part, the particle has a tendency to expand the search space, that is, they have ability to explore new area. Therefore, they acquire a global search capability by adding the first parts.

3. Adaptive Particle Swarm Optimization

APSO technique makes use of the fact that the general motion in a frame is usually coherent, i.e. if the macro block around the current macro block is moved in a particular direction then there is a high probability that the current macro block will also have a similar motion vector. This algorithm uses the motion vector of the macro block to its immediate left to predict its own motion vector.

Instead of constant step size, the authors have modified velocity and position equations of PSO to achieve adaptive step size for video compression, which are used to predict best matching macro block in the reference frame with respect to macro block in the current frame for which motion vector is found.

To get the adaptive step size, the velocity and position equations of PPSO are modified as given below. Instead of constant value of *W* and *C* as in PSO, the time varying nonlinear inertia weight (*W*) and time varying nonlinear acceleration coefficient(*C*) expressed in terms motion vectors are used for getting the true motion vector dynamically. The W and C are presented by Eq. (3) and Eq. (4)

$$W = r * Max[|X|, |Y|] \quad (3)$$

$$C = r * Min[|X|, |Y|] \quad (4)$$

X and *Y* is the x and y coordinates of the predicted motion vector. *r* is the random number between 0 to 1.

In APSO, velocity equation is expressed as Eq. (5). The parameters used in Eq. (5) are changed dynamically with each generation.

$$V_{K+1}^{i} = W * C \ * r \quad (5)$$

The velocity term in Eq (5) is added with previous motion vector to predict the next best matching block as given in Eq (6)

$$S_{k+1}^{i} = V_{k+1}^{i} + (|X| + |Y|) / |X| \quad (6)$$

4. Adaptive Particle Swarm Optimization for Video Compression

In APSO, for video compression for each block in the frame, a search is made in an earlier frame of the sequence over a random area of the frame. The search is for the best matching block viz. the position that minimizes a distortion measured between the two sets of pixels comprising the blocks. The relative displacement between the two blocks is taken to be the motion vector. Usually the macro block is taken as a square of side consists of 16 pixels. The compression ration is 128:1 or 256:2. The each block size of 16 x 16 is compressed into two pixels which are nothing but motion vectors.

In the APSO, small population i.e. five swarms are used to find best matching block. The initial position of block to be searched in reference frame is the predicted motion vector as expressed in Eq. (6). In APSO, the number of generations is taken as 2. In first step, the *pbest* is updated among five swarms and *gbest* is nothing but *pbest*. Again in second step, new *pbest* is found. The *pbest* and *gbest* are compared and *gbest* is updated. The best matching block in search space is the *gbest*. The cost required for finding best matching block or *gbest* in the reference frame is ten blocks, which is less than existing methods.

The mean absolute difference (MAD) is taken as objective function or cost function in APSO and is expressed as in Eq. (7).

$$MAD = \frac{1}{MN}[\sum_{P=1}^{M}\sum_{Q=1}^{N}|CurrentBlock(P,Q) - \mathrm{Re}\,ferenceBlock(P,Q)|] \quad (7)$$

Where, M = Number of rows in the frame and N = Number of columns in the frame. The objective quality obtained by APSO has been measured by the peak signal-to-noise ratio (PSNR), which is commonly used in the objective quality comparison. The performance of the proposed method is evaluated by following Eq (8)

$$PSNR = 10\log_{10}\frac{255^2}{\frac{1}{MN}\sum_{P,Q=1}^{M,N}(OrigionalFrame(P,Q) - CompensatedFrame(P,Q))^2} \quad (8)$$

A further small improvement in the APSO is to check for zero motion prejudgment. If current macroblock matches with macroblock in the reference frame i.e. cost is zero then motion vector are directly stored as zero motion vector instead of gaining the motion vector through APSO. The zero motion prejudgment saves considerable amount of computational time.

5. Results and Discussions

The presented method APSO has been tested for three medical videos as shown in Figure 1 to 3. The performance of the proposed method is measured in terms of the average peak signal to noise ratio (*PSNR*) and percentage of saving in the motion estimation time. To test the efficiency of the proposed algorithm with existing methods, the algorithms are executed in single machine i.e. PIV, 3 MHz CPU, 2GB RAM and MATLAB 7.0. Video sequence with distance of two frames between current frame and reference frame are used to generate frame-by-frame results of the proposed algorithm. The performance of APSO is compared with other existing methods such as ES, DS, ARPS and the results are presented in Table 1 and Table 2. Figure 4 to Figure 6 shows the comparison of *PSNR* with other existing methods for three medical video sequences respectively. The speed of APSO is found to be faster than that of already published methods and *PSNR* is close to published methods as shown in Table 3. The APSO saves computational time up to 93.70% to 83.57% with PSNR degradation of -0.45 to -0.422 as compared to ES. Similarly, APSO saves computational time up to 48.78% to 28.42% with PSNR gain of +0.9152 to +0.6859-0.422 as compared to DS and ARPS.

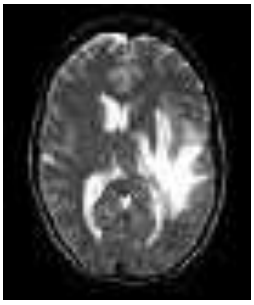

Figure 1 Original Frame of medical video CT1

Figure 2 Original Frame of medical video CT2

Figure 3 Original Frame of medical video CT3

Table 1: Comparison of average *PSNR* of APSO and existing methods

Sr. no	Type of video Sequence	No. of Frames	Average PSNR in db			
			ES	DS	ARPS	APSO
1	CT1	25	27.82	26.47	26.44	27.38
2	CT2	30	28.84	27.74	27.70	28.39
3	CT3	24	28.77	27.63	27.58	28.35

Table 2: Comparison of computational time of APSO and existing methods

Sr. no	Type of video Sequence	No. of Frames	Average PSNR in db			
			ES	DS	ARPS	APSO
1	CT1	25	4.24	1.23	1.05	0.63
2	CT2	30	5.28	1.67	1.34	0.84
3	CT3	24	4.14	1.17	0.95	0.68

Table 3: Computational time saving and *PSNR* gain by APSO over existing methods

Sr. no	Type of video Sequence	No. of Frames	Existing methods		
			ES	DS	ARPS
1	CT1	25	85.14	48.78	40
	CT2	30	84.091	49.70	37.31
	CT3	24	83.57	41.88	28.42
2	CT1	25	-0.432	+0.9152	+0.9472
	CT2	30	-0.45	+0.6434	+0.6859
	CT3	24	-0.422	+0.7191	+0.7628

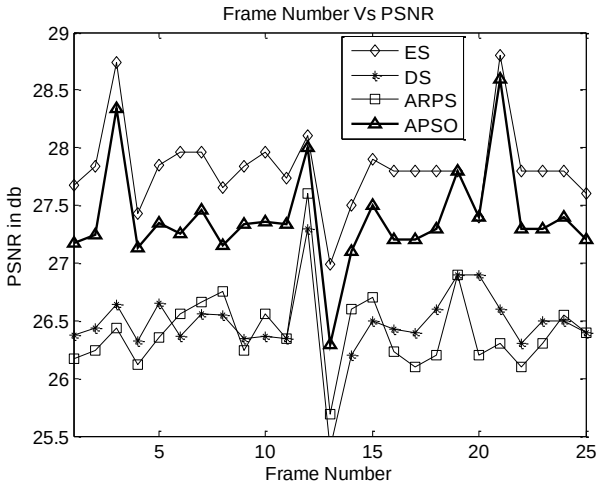

Figure 4 Comparison of *PSNR* for medical video CT1

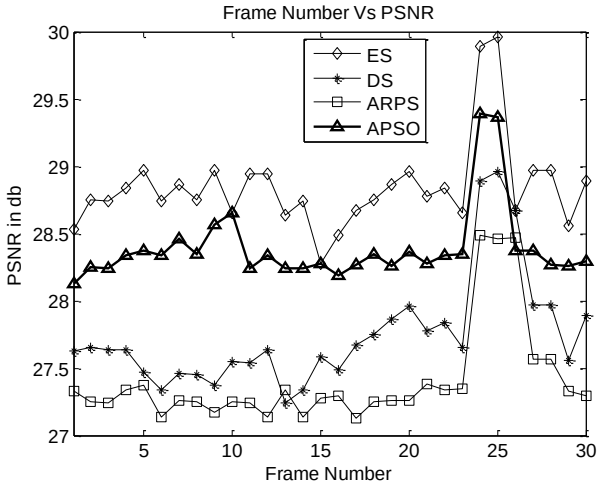

Figure 5 Comparison of *PSNR* for medical video CT2

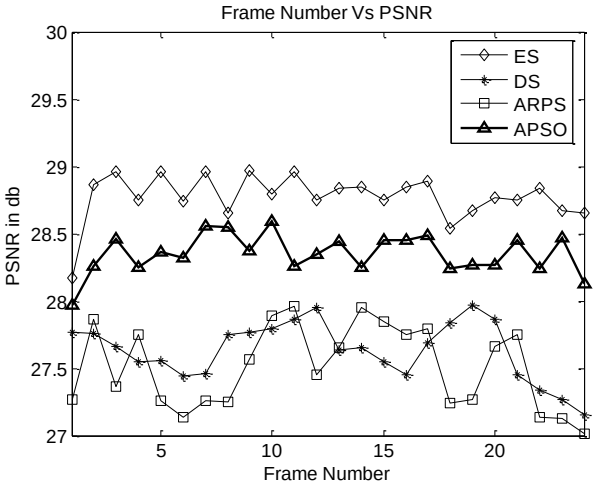

Figure 6 Comparison of *PSNR* for medical video CT3

6. Conclusion

The paper presents APSO for medical video compression. The velocity and position equations of PSO are modified to achieve adaptive step size for video compression, which are used to predict best matching macro block. The initial search in APSO start in an area where there exists high probability of finding a good matching block due to modified step equation of PSO. The results show promising improvement in terms of accuracy (*PSNR*), while drastically reducing the computational time. More than 90% of computational time saving in the video compression is achieved as compared to ES algorithm. This saving comes with only -0.45 to -0.422 db degradation in the *PSNR*. APSO saves computational time up to 48.78% to 28.42% with *PSNR* gain of +0.9152 to +0.6859 as compared to DS and ARPS. In proposed technique, zero motion is stored directly. The APSO outperforms well than conventional fast block matching methods in terms of both peak signal to noise ratio (PSNR) and computational complexity.

References

[1] Jianhua Lu and Ming L. Liou, "A Simple and Efficient Search Algorithm for Block Matching Motion Estimation," *IEEE Trans. Circuits and Systems for Video Technology*, vol. 7, no.2, pp. 429-433, 1997.

[2] Renxiang Li, Bing Zeng and Ming L. Liou, "A New Three- Step Search Algorithm for Block Motion Estimation," *IEEE Trans. Circuits and Systems for Video Technology*, vol. 4, no. 4, pp. 438-442, 1994.

[3] Lai-Man Po and Wing –Chung Ma, "A Novel Four- Step Search Algorithm for Fast Block Motion Estimation," *IEEE Trans. Circuits and Systems for Video Technology*, vol. 6, no.3, pp. 313-317, 1996.

[4] Shan Zhu and Kai-Khuang Ma, "A New Diamond Search Algorithm for Fast Matching Motion Estimation," *IEEE Trans. Image Processing*, vol. 9, no.2, pp. 287-290, 2000.

[5] Yao Nie and Kai-Khuang Ma, "Adaptive Rood Pattern Search for Fast Block-Matching Motion Estimation,"

IEEE Trans. Image Processing, vol.11, no.12, pp. 1442-1448, 2002.

[6] M. J. Tsai, J. D. Villasenor, and B. K. T. Ho, "Coronary angiogram video compression," in *Proceedings of 1994 IEEE Nuclear ScienceSymp. Medical Imaging Conf.*, Norfolk, VA, Oct. 1994, paper no. 8M65.

[7] D. Gibson, G. Tsibidis, M. Spann, and S. Woolley, "Angiogram video compression using awavelet-based texture modeling approach," in *Proc. 2001 British Conf. Medical Image Understanding and Analysis*, Birmingham, U.K., Jul. 2001.

[8] S. B. Gokturk, C. Tomasi, B. Girod, and C. Beaulieu, "Medical image compression based on region of interest, with application to colon CT images," in *Proc. 23rd Annual Int. Conf. IEEE Engineering in Medicine and Biology Society*, Istanbul, Turkey, Oct. 2001.

[9] R. C. Eberhart and Y. Shi, "Comparison between genetic algorithm and particle swarm optimization," *IEEE Int. Conf. Computt. Anchorage*, AK, pp. 611-616, May 1998,

[10] R. Eberhart and Y. Shi, "Special issue on particle swarm optimization," *IEEE Trans. Evol. Comput.* vol. 8, no.3, pp. 201–228, 2004.

[11] M. Senthil Arumugam, M. V. C. Rao and Alan W.C. Tan, "A new novel and effective particle swarm optimization like algorithm with extrapolation technique," *International Journal of Applied Soft Computing, Elsevier*, vol. 9 , pp. 308-320, 2009.

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