1st NATIONAL CONFERENCE ON INTELLIGENT COMPUTING & INFORMATION TECHNOLOGY (NCICIT-2013)

NCICIT

2013

November 21, 2013 CUET, Chittagong, Bangladesh

Department of Computer Science & Engineering

Chittagong University of Engineering & Technology (CUET)

Chittagong-4349, Bangladesh

www.cse.cuet.ac.bd/ncicit2013

NCICIT

2013

November 21, 2013 CUET, Chittagong, Bangladesh

1st National Conference on Intelligent Computing & Information Technology

Conference Digest

Editors

Dr. Mohammed Moshiul Hoque
Mir. Md. Saki Kowsar
Kazi Zakia Sultana

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Department of Computer Science & Engineering
Chittagong University of Engineering & Technology (CUET)
Chittagong-4349, Bangladesh

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Welcome Address from Organizing Chair & Secretary

On behalf of the Organizing Committee we welcome you to the 1st edition of National Conference on Intelligent Computing & Information Technology (NCICIT 2013). This conference is being organized by the Department of Computer Science & Engineering, Chittagong University of Engineering & Technology. The NCICIT conference addresses the interdisciplinary and multidisciplinary research in intelligent computing and information technology with artificial intelligence, intelligent agents, computer vision, pattern recognition, natural language processing, assistive technology, and many more. We invite participation and encourage discussion and sharing of ideas across a diverse audience.

NCICIT 2013 has a specific focus on Recent Trends in Intelligent Computing and Information Technology, which requires advancement of the start-of-the-art in the empirical, algorithmic, mathematical, social, and engineering aspects of intelligent system in an integrated manner. A joining of the discipline is essential for enabling systems to help people in their efforts to be more productive and to enjoy a high quality of life.

Full papers submitted to the conference were thoroughly reviewed and discussed. A total of 37 standard presentations have been selected from about 67 submissions received. All publications were reviewed from at least two independent reviewers and a technical committee member in order to ensure high quality of contributed material as well as adherence to conference topics. Accepted papers will be published in a conference e-proceeding. We plan to edit a special issue in the CSERJ (Computer Science & Engineering Research Journal). Presented papers with reasonable extension will be invited for submission in this issue.

The conference could not have occurred without the extensive volunteer effort put forth by the advisory committee, organizing committee, technical committee, publication committee, finance committee and reviewers. We would like to thank the distinguished guests, and keynote speakers for their participation and attendance. We would like to express our appreciation to IEB, Chittagong for technical co-sponsorship and Confidence Cement, RSRM, National Bank Ltd., BTCL, and Cefalo for their cooperation.

We would like to express our gratefulness to the Prof. Dr. Jahangir Alam, Vice-chancellor, CUET and Prof. Rafiqul Alam, Pro Vice-chancellor, CUET for their continuous support, encouragement, and innovative ideas. Finally, we are especially thankful for the hard work by authors who submitted papers. We hope that all participants will find the conference enjoyable, informative, and thought-provoking.

We hope you will really enjoy the 2013 edition of NCICIT and your stay in CUET.

Dr. Kaushik Deb
Organizing Chair, NCICIT 2013

Dr. M. Moshiul Hoque
Organizing Secretary, NCICIT 2013

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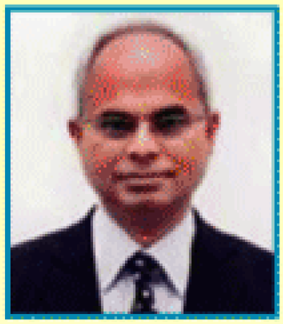

Message

I really appreciate that the Department of Computer Science & Engineering, Chittagong University of Engineering & Technology (CUET) is organizing the 1st National Conference on Intelligent Computing and Information Technology (NCICIT 2013) at its own campus.

Ministry of Information & Communication Technology of Bangladesh is working relentlessly to make the Vision 2021: Digital Bangladesh true and a success through ensuring Information & Communication Technology as a part of our national culture. We have mainstreamed ICT to empower citizens in eradicating poverty and establishing good governance by delivering better citizens services through promoting quality education, healthcare and law enforcement for all. The country is going forward in full swing by the effective utilization of Information & Communication Technology for positive change in society and for balanced socio-economic upliftment. We hope the conference will open up a new door to the young researchers of our country and thus smoothen the way ahead towards implementing the ultimate mission and vision of the country for its people.

University is a platform for creating and discovering knowledge and conference paves the way towards strengthening the platform for researchers, scientists, engineers, and practitioners throughout the country. Their latest findings and innovations in relevant fields make the way of solving national problems easier and smoother. Moreover, it certainly makes an opportunity to the researchers for knowledge sharing and builds a framework for joint research collaborations within the research communities. I believe that this event would promote greater corporation among the young researchers of the country through networking of the individuals and practitioners engaged in the respective area.

I wish a grand success of the conference.

Md. Nazrul Islam Khan
Secretary

Chittagong University of Engineering & Technology (CUET)
Chittagong-4349, Bangladesh

Message

I start with my utmost gratitude to the Almighty Allah for giving us this precious opportunity to gather at this memorable event of the 1st National Conference on Intelligent Computing and Information Technology (NCICIT 2013) with the theme: Recent Trends in Computing and Information Technology. This conference is a step towards achieving the vision of exploring new ideas and concepts in the arena of computer science and developing the practice of research and innovations in the South East region of the country. This conference will be undoubtedly a good starting point to interchange knowledge and skills in the area of Computer Science. We are looking forward to unhide digital solutions and to comprehend and contribute to local and global needs.

It is a great pleasure for us to welcome all delegates and participants to this conference. I would like to congratulate the Department of Computer Science & Engineering, CUET for their outstanding drive in successfully organizing this conference. I believe that this occasion will provide a common platform for researchers, scientists, engineers, and practitioners throughout the country to present their latest findings as well as motivate to come up for effectively utilize them in solving many practical problems. It will certainly strengthen relationships among the researchers through knowledge sharing and providing the necessary drive in joint research collaborations within the research communities. I dream that today's attempt of the department of CSE for this conference will be strong foundation towards a better tomorrow.

Last of all, I would also like to thank all the conference sponsors and collaborators. I am confident that your continued support and interest will make our attempt and dream of making CUET a top class university possible and true. I hope that we will be able to make the significant inroads to turning outlined mission and vision of the department of CSE, CUET into reality through future conferences.

I wish a grand success of the conference.

Prof. Dr. Md. Jahangir Alam
Vice-Chancellor

Chittagong University of Engineering & Technology (CUET)
Chittagong-4349, Bangladesh

Message

I am delighted that the Department of Computer Science & Engineering of Chittagong University of Engineering & Technology is going to arrange the 1st National Conference on Intelligent Computing and Information Technology (NCICIT 2013). We know that researches and developments in Computer Science pave the way to building a technologically developed country and world as well as to progress in changing the quality of life. This conference is the first attempt from the Department of CSE to address the recent challenges and solutions in this field.

I think the conference has not only made a platform of exchanging the views among the expert researchers, practitioners, academicians of the country but also will certainly go a long way in enriching the knowledge of the participants. I appreciate the whole-hearted commitment and involvement of the organizers who took the attempt of this huge conference. I also thank all the sponsors for their effort to encourage academic research by way of their sponsorships. I hope all the organizers and collaborators will come forward to host more of such national and international conferences in future.

I sincerely hope and believe that this conference will encourage the research potential of our young students and researchers of this region as well as persuade them to build a technologically developed world through innovative contribution in information technology.

I wish the conference a grand success.

Professor Mohammad Rafiqul Alam
Pro-Vice Chancellor

Faculty of Electrical & Computer Engineering
Chittagong University of Engineering & Technology (CUET)
Chittagong-4349, Bangladesh

Message

I am really happy that the Department of Computer Science & Engineering of Chittagong University of Engineering & Technology is organizing the 1st National Conference on Intelligent Computing and Information Technology (NCICIT 2013). The conference is covering the significant and diverse areas of Computer Science for growing attractions to the researchers and practitioners while keeping abreast the recent trends in this arena to come up with new solutions for effective use in the context of our country. It is my great pleasure to welcome all of you, the research community to this conference and I believe your presence will encourage young researchers, students and practitioners in this field.

Grounded in our commitment to engineering education innovation and interdisciplinary research, we offer our students a rich educational experience, an experience that provides intellectual rigor and cross-disciplinary breadth in an intimate, student-centered environment. It is a good time for CSE to be thoughtful, to be aggressive, to think big, reach high, and become even greater as a department. I hope that the conference will open up door towards our students and researchers to reach their goals as well as step forward to the greater researcher community of the world in this area. I look back with satisfaction on our successes and look forward with anticipation to even more successes in the future. Department of CSE, CUET has always urged its students and alumni to dare to dream big dreams and not to set limits on their potential.

Before I end, I would like to thank our sponsors, and the committee members of this event for their outstanding drive in making this conference of high magnitude a success. I would also like to wish you a pleasant stay at CUET and having a fruitful conference.

Thank you.

Prof. Dr. Muhammad Ibrahim Khan
Dean

Department of Computer Science & Engineering
Bangladesh University of Engineering & Technology (BUET)
Dhaka, Bangladesh

Message

I have the pleasure to congratulate all my colleagues at CUET for taking the initiative of organizing the 1st National Conference on Intelligent Computing & Information Technology (NCICIT 2013) at its great campus. For the last quarter century we have been talking about fate-changing potential of the technology that made Bill Gates the richest man on earth from scratch in 15 years, without making any mark at all while less-educated, underprivileged, sufficiently ignored, undercared girls of the country have been earning invaluable foreign exchanges for the country, and ill-fated low skilled people are sending so much of remittances from the Middle East to inject lifeblood to our economy. We, the educated section of the country, should now come up to prove our worth to the development of our country. In line with the present Government's commitment to creation of Digital Bangladesh, I understand, CUET has been selected as a site for ICT Incubator. I am sure the enormous zeal of faculty members and students alike CUET, will make no stone unturned to make it the first success story of ICT initiatives for the country.

With the introduction of so many gadgets into our everyday life it is becoming more and more a necessity for us to introduce devices for intelligent interaction with such systems. The world is being flooded with information and misinformation at much accelerated a rate than ever before as a result of which it has become impossible to process them manually in time to obtain necessary information. Only effective way of handling them is to use intelligent systems, expert systems, artificial intelligence. Being a country that facilitates existence of the most important ingredient of human race, the humans, using only 1/24th part of the average land mass and resources used by for an average world citizen, we need to be able to optimally utilize the scant resources ensuring the highest productivity with the help of versatile ICT technologies.

I am sure that the deliberations of scientists, academicians and practitioners of the discipline will take into account these significant concerns of our country and will come up with recommendations that will drive us to the right direction in the field of ICT. I sincerely believe that the NCICIT 2013 will put stress on the innovations and effective utilization of ICT in national context and prove to be a successful forum for the exchange of ideas of research, innovation and practice.

Professor Dr. M. Kaykobad

Institute of Engineers, Bangladesh
Chittagong Centre

Message

First of all, I would like to extend our warmest appreciation to the Department of Computer Science & Engineering (CUET) for arranging the National Conference for the first time at its own campus. We are really excited about the participation and scope of the technical program that the 1st National Conference on Intelligent Computing and Information Technology (NCICIT 2013) will offer to the relevant arena of information technology being organized by the Department of Computer Science & Engineering (CUET). I truly believe that NCICIT 2013 promises to be a technical program addressing the critical problems and needs of our country and the application of Computer Science for solving the problems in flair of less complexity.

The Institution of Engineers, Bangladesh aims to promote and advance the science, practice and business of engineering in all its branches throughout Bangladesh and abroad as well as encourage original researches and diffusion of knowledge through holding and collaborating the conferences and other literary activities. I really hope that NCICIT 2013 will be able to make another step forward towards implementing the mission of IEB. This event will bring together the entire community of computer science researchers into one large venue and provide them opportunity to share their technical knowledge and views in context of the South East region of the country.

This conference, with its top-notch technical program and exciting social events, reflects the tireless effort of many individuals. I hope the conference is going to be a platform for making their contributions a reality and thus open up a space to the young researchers for exercising new knowledge in the area of Computer Science. Finally, I would like to thank all researchers, practitioners, and students who are presenting their works in this conference and also to the Department of CSE, CUET for their effort to make the event an ultimate success.

I wish you a productive conference, and hope you enjoy your time in NCICIT 2013.

Engr. Manzarey Khorshed Alam
Chairman

Department of Computer Science & Engineering
Chittagong University of Engineering & Technology (CUET)
Chittagong-4349, Bangladesh

Message

I am really privileged one as the department of CSE, CUET nominated me to the key role of publication for the NCICIT-2013. For the first time of organizing technical conference by the department, it got unyielding support from the CUET administration. On the other hand, we received overwhelming responses not only from the local researchers, it is also from abroad. At present, through extensive reviews, we got 67 number of papers to be published in the conference proceedings. We divided them into four categories: (a) Intelligent Systems & Computer Vision, (b) Algorithms, Operating Systems & Data Mining, (c) Communication, Information Security, & Computer Networks, and (d) Web Based Applications & Bioinformatics.

Our conviction is that the conference would be immensely beneficial to create conducive environment for research and innovation, especially, among young scientists of the country as well as the university, CUET.

Last but not least, the conference really would not have been in this position, if there is not adequate support from our valued sponsors and advertising organizations. We are sincerely grateful to their generous patronization.

Finally, we look forward to have vibrant participation of all concerned for an ever successful conference, NCICIT-2013.

Mir Md. Saki Kowsar
Chair, Publication Committee

We wish NCICIT-2013 a success

With best compliments from

Keynote Speaker

How to Write Good Research Papers in Computer Science

Professor Dr. Md. Saidur Rahman
Department of Computer Science and Engineering,
Bangladesh University of Engineering and Technology (BUET),
Dhaka-1000, Bangladesh
E-mail: saidurrahman@cse.buet.ac.bd

Abstract:

In this talk I will present some rules and conventions for writing research papers in theoretical computer science. We write research papers to publish our research results. For doing research in theoretical computer science, one may follow the following steps. Explore your area of interest and choose a research problem. It is recommended to find a group of two or three co-researchers to work on the problem. Read papers related to the problem published in good journals and present the paper by turn in the group. Sit frequently for brainstorming on the problem and try to find non-trivial results. Present your works in informal workshops and local conferences for getting feedback, and then write your results as a good paper for publishing in an international conference and finally publish the paper in a reputed journal.

A research paper in theoretical computer science has the following components: title, authors name and affiliation, abstract, keywords, introduction section, preliminaries section, one or more sections describing main results, conclusion and references. Abstract is the full paper written within ten lines whereas Introduction is the full paper written in 2-3 pages. To make the paper self-contained Preliminaries section plays an important role. Relevant notations, definitions, previous and preliminary results are given in this section. Main results of the paper are presented using one or more sections as required. Interesting implications of results and related open problems are presented in conclusion section. A standard and consistent style should be followed for writing reference entries. If needed acknowledgement is inserted after conclusion and appendices are added at the end of the paper. Maintaining a smooth logical flow throughout the paper is very much essential.

One should know the proper way of citing works of others. If we need to describe works of others for presenting our results then we should write those in our own words using proper references. Note that uncredited verbatim copying of a portion of a paper is treated as plagiarism, and we should be very careful about it. It is very important to choose good conferences and journals for publishing research results. One publication in a good journal is worthier than publishing dozens of papers in so-called journals. We should remember that a publication in a bad journal is enough to spoil the career of a researcher. Thus we should not entertain the requests that we often receive through emails from so-called journals which accept papers within a few days or even before submitting it!

Short Biography:

Mr. Saidur Rahman received B. Sc. Engg. degree in EEE in 1989 and M. Sc. Engg. degree in CSE in 1992 from BUET. He received M. Sc. and Ph. D. degrees in Information Sciences from Tohoku University, Japan in 1996 and 1999, respectively. Professor Rahman started his career as an Assistant Engineer at ICS, AERE in 1990. He joined the CSE Department, BUET, as a Lecturer in 1991. He is currently serving the department as a Professor. He also worked as an Associate Professor at Graduate School of Information Sciences, Tohoku University, Japan during 2003-2004. Professor Rahman specialized in theoretical computer science and researches on graph algorithms, graph drawing algorithms, VLSI physical layout algorithms, computational geometry, internet routing protocols and bioinformatics. He has developed several efficient algorithms for planar graph drawings which have been successfully used in applications like circuit schematics, VLSI floorplanning and architectural floorplanning. He has around 100 publications in internationally reputed journals and conferences. He served as a program committee member of International Symposium on Graph Drawing (GD) in 2003, 2005 and 2007, International Symposium on Algorithms and Computation (ISAAC) in 2006 and 2011, Annual Meeting of Asian Association for Algorithms and Computation (AAAC) in 2010, 2011 and 2012, and IEEE Pacific Visualization Symposium (PacificVis) in 2012.

Keynote Speaker

Evolutionary Algorithms for Many-objective Optimization Problems

Professor Dr. Md. Monirul Islam
Department of Computer Science and Engineering,
Bangladesh University of Engineering and Technology (BUET),
Dhaka-1000, Bangladesh
E-mail: saidurrahman@cse.buet.ac.bd

Abstract:

Many real world problems involve simultaneous optimization of several incommensurable and often competing objectives. The presence of multiple objectives in a problem, in principle, gives rise to a set of optimal solutions (commonly known as Pareto-optimal solutions), instead of a single optimal solution. In the absence of any further information, one of these Pareto-optimal solutions cannot be said to be better than the other. This demands a user to find as many Pareto-optimal solutions as possible by executing an algorithm only one time. Evolutionary algorithms are suitable for such problems because of their ability to find multiple Pareto-optimal solutions in one simulation run. However, these algorithms face several difficulties when optimization problems have three or more objectives. This type of problems is termed as many-objective optimization problems (MaOPs). Solving MaOPs is very important, because real-world problems intrinsically have several objectives. From a practical point of view, it is often desirable with most applications to include as many objectives as possible. We will discuss here problems, challenges and ways to solve MaOPs using evolutionary algorithms.

Short Biography:

Mr. Monirul Islam received the B. Sc. Engineering degree from the Khulna University of Engineering and Technology (KUET), Khulna, Bangladesh, in 1989, the M. Sc. Engineering degree from the Bangladesh University of Engineering and Technology (BUET), Dhaka, Bangladesh, in 1996, and the Ph.D. degree from the University of Fukui, Fukui, Japan, in 2002.

He is currently a Professor in the Department of Computer Science and Engineering, BUET. He also worked as a Visiting Associate Professor at the University of Fukui, Japan during 2007-2009. He has been an invited as a keynote speaker at several international conferences and has more than 100 refereed research publications in evolutionary computation, neural network and datamining. His research interests include evolutionary computation, neural networks, global optimization, and data mining. In addition to basic research, he works closely with many industrial partners on various real-world problems.

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Oral Presentation Guidelines

NCICIT 2013 - INSTRUCTIONS FOR SESSION CHAIRS

Session Chairs are asked to kindly review the conference program before their session.

All rooms will be equipped with a video projector, speakers, and a screen. Authors have been asked to use NCICIT laptops/pcs and to meet their Session Chair at the session room a few minutes before the beginning of their session to check equipment or ppt version compatibility. If something is not working properly, please ask for help to one of the Conference Helpers.

IMPORTANT NOTE: some parallel sessions run in sequence with no break.

The length of each oral presentation is restricted to 15 minutes, including questions. The authors should be strongly advised to keep their oral presentation within 12 minutes, and to leave 3 minutes for discussion with the audience and change of speaker.

To ensure that the conference program remains on schedule, Session Chairs are asked to kindly enforce the time limits on each presentation. Please, try to respect the schedule.

NCICIT 2013 - INSTRUCTIONS FOR PAPER PRESENTERS

Each paper is allocated a 15-minute presentation slot. Please limit your presentations to 12 minutes, to allow sufficient time for Q&A and speaker transition. Each room is equipped with an LCD projector. Authors are asked to meet their Session Chair at the session room a few minutes before the beginning of their session. Presenters are asked to bring their own slides and copy to the NCICIT laptops/pcs and to check for equipment or ppt version compatibility before beginning of the session. If something is not working properly, please ask for help to your session chair or to one of the conference/sessions helpers.

Social Events

Opening Session
Thursday, November 21, 9:00 AM
All registrants are invited to join the Opening session. This event will be inaugurated by the honorable Vice-Chancellor Prof. Dr. Jahangir Alam, CUET. The Inaugural ceremony will start from 9:00 am at the CUET Central Auditorium.

Breaks
Thursday, November 21, 11:10 AM~11:30 AM & 03.15PM~3:30PM
Complementary coffee and light refreshments will be provided during the scheduled breaks in the Department of Computer Science & Engineering.

Inauguration Ceremony
Thursday, November 21, 11:30 AM
All the participants and invited guests are requested to attend this ceremony. Mr. Md. Nazrul Islam Khan, Honorable Secretary, Ministry of Information and Communication Technology, Government of People's Republic of Bangladesh, has given his kind consent to grace this event as the Chief Guest. Prof. Dr. Md. Jahangir Alam, Vice Chancellor, CUET, Prof. Md. Rafiqul Alam,Pro Vice-chancellor, CUET and Ms. Hosne Ara Begum, Managing Director, Bangladesh Hi-Tech Authority will join the program as Special Guests.

Lunch
Thursday, November 21, 1:00 PM
After closing the first technical session, a lunch will be offered in the Department of Computer Science & Engineering (CSE).

Closing Session
Thursday, November 21, 5:30 PM
After closing the last technical session, a closing session will be held in the CUET Central Auditorium.

Cultural Evening
Thursday, November 21, 6:00 PM
All the registrants are invited to attend the exquisite Cultural Evening. Students of CSE discipline of CUET will be performing in this event during the Banquet.

Banquet
Thursday, November 21, 6:00 PM
All the registered participants and invited guests are requested to join in the Banquet at the CUET Central Auditorium. There will be a special token for entry in the banquet hall.

Internet Access
During the NCICIT 2013, participants will be able to connect their PC, tablet, or smart phone with Internet through wireless network.

For further information, please ask the conference staff at the Registration Desk.

Final Program

Time	Events		Location
08:00-09:00	Registration Open & Conference Kit Collection		Dept. of CSE
09:00-09:30	**Opening Session**		Dept. of CSE
09:10-11:10	**Technical Session 1A: Intelligent Systems & Computer Vision** **Session Chair: Prof. Dr. Md. Tazul Islam, Dept. of ME, CUET** **Session Co-chair: Dr. Assaduzzaman, Dept. of CSE, CUET**		MM Lab, Dept. of CSE
	P137	Walking Steps Counting and Distance Measurement on User Smart Phone	
	P153	Mobile Robot Controlled by Voice Command based on GCM Technology using Android Device	
	P175	Development of Digital Weighting Machine using 89S52 Microcontroller Architecture	
	P136	Content Based Image Retrieval using Combined Features	
	P143	Shadow Detection and Removal from Single Image using YC_bC_r Color Space	
	P144	Face Detection using RGB Color Model	
	P151	A Method for Pitch Detection of Speech Signal in Nosiy Environment	
	P165	Analysis of Normal and Infected Bio Cell by using Dual Nanoprobe	
	Technical Session 1B: Algorithms, Operating Systems & Data Mining **Session Chair: Dr. Md. Abul Kashem, Dept. of CSE, DUET** **Session Co-chair: Dr. Md. Shamsul Arefin, Dept. of CSE, CUET**		Room: L4, Dept. of CSE
	P121	A Novel Approach of Multilevel Feedback Queue Scheduling	
	P125	Irregular Total Labeling of Knodel Graphs	
	P148	Analyzing Progressive BKZ Lattice Reduction Algorithm	
	P167	An Efficient Page Replacement Algorithm	
	P168	A Modified Approach for Selecting Optimal Initial Centroids to Enhance the Performance of K-Means Algorithm	
	P128	Exploiting GPU Parallelism to Optimize Real World Problems	
	P139	Sentiment Mining in Social Network using Textual Opinion	
	P173	Dimensionality Reduction and Cluster Center Selection: An	

11:10-11:30	**Tea Break**		Dept. of CSE
11:30-12:45	**Inauguration Ceremony**		Central Auditorium
12:45-2:30	**Prayer & Launch**		Dept. of CSE
02:30-03:15	**Keynote Session** **Session Chair: Prof. Dr. Md. Ibrahim Khan, Dept. of CSE, CUET**		Central Auditorium
	Evolutionary Algorithms for Many-objective Optimization Problems Prof. Dr. Md. Monirul Islam, Bangladesh University of Engineering & Technology (BUET)		
03:15-3:30	**Tea Break**		Dept. of CSE
03:30-05:30	**Technical Session 2A: Communication, Information Security, & Computer Networks** **Session Chair: Dr. Abdul Matin Bhuiyan, Dept. of EEE, CUET** **Session Co-chair: Dr. AFM Mirza Rashedul Hasan, Dept. of ICE, RU**		Room: L4, Dept. of CSE
	P140	IPv6 Deployment in Bangladesh: An Analysis of the Present State and the Way Forward	
	P149	User Authentication Approach for Data Security between Smartphone and Cloud	
	P150	A Binary Code Lock System	
	P172	Design an Efficient MAC Protocol for Cognitive Radio Systems using Game Theory	
	P179	An Efficient Approach to Save Cluster Head Energy of Sensor Network	
	P180	Software Defined Radio for PC to PC Data Communication	
	P184	Linear Polarization Switchable Patch Array Antenna using magic-T Bias Circuit and Orthogonal Feed	
	Technical Session 2B: Web Based Applications & Bioinformatics **Session Chair: Prof. Dr. Md. Shahadat Hossain, Dept. of CSE, CU** **Session Co-chair: Dr. Quazi Delwar Hossain, Dept. of ETE, CUET**		
	P166	Development of Asynchronous Replication Model for Heterogeneous Environment	
	P169	A Sequence Alignment Algorithm and Tools for Molecular Replacement	
	P170	Performance of Warshal Algorithm and Dynamic Programming for Markov Chain in Local Sequence Alignment	
	P171	A New Approach of Disk Scheduling Algorithm	
	P124	e-Government and Its Implementation Challenges in Nepal	
	P130	An Integrated Online Courseware Design Approach	
	P146	Intelligent Decision System for Evaluating of Job Offers	
	P176	Blended Learning Approach for Engineering Education-An Improvement Phase of Traditional Learning	
05:30-06:00	Closing Ceremony		Central Auditorium
06:00-08:00	Cultural Evening & Banquet		Central Auditorium

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Technical Session 1A 09:10~11:10	MM Lab, Dept. of CSE
Chair: Prof. Dr. Md. Tazul Islam	CUET
Co-chair: Dr. Assaduzzaman	CUET

P 137: Walking Steps Counting and Distance Measurement on User Smart Phone

N. Jahan PSTU
M. A. Masud PSTU
N. J. Bubly PSTU
L. Khatun PSTU

As smart phone proliferates in our society all over the world, day by day it is becoming more familiar tool that is highly adapted in our everyday life. Most importantly these smart phones are equipped with various sensors that can be utilized to build up a wide variety of applications. In this paper, we present a method for counting the number of steps of a smart phone user, while walking at any variable speed. Hence, the cellular device must be configured with the acceleration and orientation sensor. For this purpose, the steps are detected based on a gravity with respect to time and the calculation of gravity is done from raw data of those sensors. The user's walking motion is recognized by android sensor (acceleration and orientation) and generated raw data. The walking distance with average step length 2.5 feet is assigned from survey study.

P 153: Mobile Robot Controlled by Voice Command based on GCM Technology using Android Devicce

S. A. Alam AUST
F. Rashid AUST
M. T. Rahman AUST
S. Islam AUST
N. J. Lisa AUST

Mobile robot controlled by voice command means a robot that can be operated by our human voice. We have used voice recognition software to identify the appropriate command for robot to responds and another android device at the command sending end to fulfill our need. In a form of a short message a robot is capable of receiving the command and performs the predefined task according to the command. A micro-controller is added in the system to read the command given from the android device and perform the task accordingly such as move forward, backward, left, right and stop. We are using cloud messaging technique using GCM technology. Because of using the GCM technology our robot can be controlled even without being in front of the robot. We can give command sitting far away from the robot using internet. PHP server is used here for the 3rd party server.

P 175: Development of a Digital Weighing Machine Using 89S52 Microcontroller Architecture

G. Mostafa AUST

In a Digital Weighing Machine (DWM), the system goes through the steps (i) acquire weight using a precise load cell, (ii) normalizing the acquired raw weight by the gain and offset of the input subsystem, (ii) acquire rate, (iii) compute cost, and (iv) show the weight, rate and cost on display unit. This paper presents the implementation methodology of these steps using real hardware. A prototype meter was emulated using 89S52/HX710 architectures and MicroTalk-8051 Learning/Dev. The proposed system was found to work within specifications.

P 136: Content-Based Image Retrieval using Haar Wavelet Transform

M. S. Iqbal CUET
M. I. H. Sarker CUET
M. I. Khan CUET

Content-based image retrieval (CBIR) system, also known as query by image content (QBIC), is an image search technique to retrieve relevant images based on their contents. Here contents refer to the color, texture and shape of the image. In this paper we propose a content-based image retrieval system, where we use Haar wavelet transform to extract the image feature. We use f-norm theory to reduce the dimension of the feature vector and finally we use Canberra distance to calculate the distance between query image and database images. Our experiment result reflects the importance of Haar wavelet transform in CBIR system.

P 143: Shadow Detection and Removal Based on YCbCr Color Space

A. H. Suny CUET
K. Deb CUET
P. Biswas CUET
M. M. Hoque CUET

Shadows in an image may reveal information about the objects shape, orientation and even about the light source. Thus shadow detection and removal is a very crucial and inevitable task of some computer vision algorithms, such as segmentation, object detection and tracking. This paper proposes a simple method to detect and remove shadows from shadow images using YCbCr color space. An approach based on statistics of intensity in YCbCr color space is proposed for detecting shadows. After the shadows

are identified, the shadow density model is applied. According to shadow density model, the image is segmented into several regions that have the same density. Then, the shadows are removed by relighting each pixel in YCbCr color space and correcting the color of the shadowed regions in RGB color space. The most salient feature of our proposed method is that after removing shadows, there are no harsh transition between the shadowed parts and non-shadowed parts and all the details in the shadowed regions remain intact.

P 144: Face Detection using RGB Color Model

B. L. Dewanjee	CUET
T. Chowdhury	CUET

Face is our primary focus of attention for conveying identity. Human face detection by computer systems has become a major field of interest. Detection of faces in a digital image has gained much importance in the last decade, with application in many fields. RGB color model is used in skin color segmentation to separate the human skin pixels. The specific values of red, green and blue components for human skin are described in this research paper. Filtering & labeling of an image and some morphological operations are used to find out face candidate. The different geometrical properties of human face are used to reject the non-human face region. Specific advantages of this approach are that skin color analysis method is simple and powerful, and the system can be used to detect multiple faces. This face detector has been applied to several test images, and satisfactory results have been obtained.

P 151: A Method for Pitch Detection of Speech Signal in Noisy Environment

S. A. Rumy	RU
M. A. F. M. R. Hasan	RU
R. Yasmin	RU
M. S. Rahman	SUST

An efficient pitch detection algorithm is proposed in this paper. The algorithm is based on time domain pitch detection algorithm. In our proposed method, instead of the original speech signal, we employ its center clipping signal for obtaining the autocorrelation function and this function is weighted by the reciprocal of the average magnitude difference function for pitch detection. The performance of the proposed pitch detection method is compared in terms of gross pitch error with the other related method. A comprehensive evaluation of the pitch estimation results on male and female voices in white noise show the superiority of the proposed method over the related method under low levels of signal to noise ratio (SNR).

P 165: Analysis of Normal and Infected Bio-cell by Using Dual Nanoprobe

M. A. Hossain	CUET
M. H. Ullah	CUET
M. A. Ullah	CUET

Knowledge of nanoprobe based bio-cell analysis method can be used to diagnostically difference between healthy and infected bio-cells. This method is made possible by using nanotechnology, a new field of science that provide a technology for human to interact with nanoscale life form organism specially cell. The electrical behaviors of healthy and infected cells are different. This paper analyses yeast cell, liver cell, and blood cell in both healthy and infected conditions to observe the differences in electrical behaviors. A dual nanoprobe is used for supplying electrical power from source to bio-cell. The voltage can be applied by two ways across the cells. One is penetrating the cell wall and another is keeping the nanoprobe in closed contact with the cell membrane. After simulation the current was measured about 2.7 times larger for liver tumor cell than healthy liver cell including cell membrane. The current flow through the healthy cell is 1.9nA whereas the current flow through a dead cell is 34pA. It is expected due to the conductivity of cytoplasm of healthy cell is greater than that of dead cell. The current is measured for a leukemia affected cell is 21.2nA. It is 2% less than the current for a white blood cell.

Technical Session 1B	**Room: L4, Dept. of**
09:10~11:10	**CSE**
Chair: Dr. Md. Abul Kashem	DUET
Co-chair: Dr. Md. Shamsul Arefin	CUET

P 121: A Novel Approach of Multilevel Feedback Queue Scheduling

O. Faruque	DUET
M. N. Akhtar	DUET
M. S. Islam	DUET

Multilevel feedback scheduling is a kind of process scheduling mechanism where process doesn't come with any priority. According to the CPU burst needed by the process and the CPU burst remaining the processes are shifted between queues of the feedback scheduler to get completed. In multilevel feedback queue the total architecture is divided into multiple prioritized queues. In each queue the processes get access of CPU for a fixed time period end after this access time the process is shifted to the next low priority queue with remaining CPU burst time. In this paper we propose a new multilevel feedback queue scheduling technique where based on the remaining CPU burst time we either execute the process immediately or shift the process to next queue. In comparison to other types of MLFQs the performance of the proposed scheduling technique is better and practical according to the consequence.

P 125: Irregular Total Labeling of Knödel Graphs

K. M. M. Haque SIU

The total edge irregularity strength *tes(G)* and total vertex irregularity strength *tvs(G)* are invariants analogous to irregular strength *s(G)* of a graph *G* for total labellings. Bača et al. [1] determined the bounds and precise values for some families of graphs concerning these parameters. In this paper, we show the exact values of the total edge irregularity strength $tes(W(3,n)) = \frac{n}{2} + 1$ total vertex irregularity strength and $tvs(W(3,n)) = (\frac{n}{4} (+ 1$ for the Knödel Graphs *W (3, n)*.

P 148: Analysing Progressive-BKZ Lattice Reduction Algoritm

M. M. Haque MU
M. O. Rahman CUET
J. Pieprzyk MU

BKZ and its variants are considered as the most efficient lattice reduction algorithms compensating both the quality and runtime. Progressive approach (gradually increasing block size) of this algorithm has been attempted in several works for better performance but actual analysis of this approach has never been reported. In this paper, we plot experimental evidence of its complexity over the direct approach. We see that a considerable time saving can be achieved if we use the output basis of the immediately reduced block as the input basis of the current block (with increased block size) successively. Then, we attempt to ⬚nd pseudo-collision in SWIFFT hash function and show that a different set of parameters produces a special shape of Gram-Schmidt norms other than the predicted Geometric Series Assumptions (GSA) which the experiment suggests being more efficient.

P 167 : An Efficient Page Replacement Algorithm

E. Kabir DUET
M. N. Akhtar DUET
M. S. Mahmud DUET

In this paper we studied different page replacement algorithms and compared their performance. We proposed a new page replacement algorithm that uses the combination of sequential and LRU methods to obtain the better performance than that of various existing methods. We also give our attention to substitute some former method characteristic based on new idea. Key concept is that the replacement algorithm should reduce the page fault of system.

P 168: A Modified Approach for Selecting Optimal Initial Centroids to Enhance the Performance of K-means

M. M. Rahman DUET
M. N. Akhtar DUET

Cluster analysis is one of the major data analysis tools to identify the behavior of data in a set of data items. The main purpose of this technique is to group data with maximum similarities into same clusters and separate data with dissimilarities into different clusters. K-means is one of the major clustering techniques that is widely used. The main problem of K-means is random selection of centroids. Clustering performance of the K-means completely depends upon the correctness of the initial centroids. In general, K-means randomly selects initial centroids which often show in poor clustering results. This paper has proposed a new approach to optimizing the designation of initial centroids for K-means clustering. We propose a modified approach for selecting initial centroids of K-means based on the sum score (ss) of the dataset. According to our experimental results the new approach of K-means clustering algorithm reduces the total number of iterations, improve the time complexity and also it has the higher accuracy than the standard k-means clustering algorithm.

P 128: Exploiting GPU Parallelism to Optimize Real-World Problems

M. H. Furhad UOU
F. Ahmed CU
M. F. Faruque UITS
M. I. H. Sarker CUET

Construction of optimal schedule for airline crew-scheduling requires high computation time. The main objective to create this optimal schedule is to assign all the crews to available flights in a minimum amount of time. This is a highly constrained optimization problem. In this paper, we implement co-evolutionary genetic algorithm in order to solve this problem. Co-evolutionary genetic algorithms are inherently parallel in nature and they require high computation time. This high computation time can be reduced by exploiting the parallel architecture of graphics processing units (GPU). In this paper, compute unified device architecture (CUDA) provided for NVIDIA GPU is used. Experimental results demonstrate that computation time can significantly be reduced and the algorithm is capable to find some good solutions in a feasible time bound.

P 139: Sentiment Mining in Social Network Using Textual Opinion

M. F. Rabby PSTU
M. A. Masud PSTU
G.M S. Islam PSTU
M. M. Billah PSTU

Opinions which are collected from social networks are the great resource for analyzing public sentiment for different purposes. Sentiment mining in social network is presented to be helpful to many sorts of people who want to know the public sentiment about their product or other thing. We propose a text analysis algorithm which calculates the polarity of textual opinions expressed in social media. This technique focuses sentiment mining in social network through step to step approach.

P 173: Dimensionality Reduction and Cluster Center Selection: An Efficient Scheme for High Dimensional dataset Clustering

M. Begum	DUET
M. N. Akhtar	DUET

In the advance technology data volume with many objects and dimensions is increasing day by day. At high dimensional dataset traditional clustering algorithms do not perform well. With increasing dimensionality some traditional algorithms produce local best possible results. There are two major issues of many partitioning clustering algorithms; one is relevant feature selection and other is selection of optimal initial clusters center. K-means is a popular clustering algorithm. But it is unable to cluster a high dimensional dataset for "curse of Dimensionality" problem .It has a major problem "to select initial centriod". In this paper, we have proposed a technique for selecting winner node of KSOM algorithm which reduces most relevant dimensions of data set. And we modified k-means algorithm for selecting initial centriod to get better cluster. When, we compared the results of proposed technique with existing techniques, our modified KSOM and K-means produce better cluster. We experimented with an IRIS data set and Milk dataset, its performance was compared with other clustering algorithm for siddheswar index, quantization errors and topographic errors.

Technical Session 2A 03:30~05:30	**MM Lab, Dept. of CSE**
Chair: Prof. Dr. Abdul Matin Bhuiyan	CUET
Co-chair: Dr. AFM Mirza Rashedul Hasan	RU

P 140: IPv6 Deployment in Bangladesh: An Analysis of the Present State and the Way Forward

F. Kabir	AUW
N. I. Mowla	AUW
S. Z. Khan	IIUC
P. Bawa	USM
A. Akhter	AUW

Internet Protocol version 6 (IPv6) is a new internet protocol, which provides more advance features than Internet Protocol version 4 (IPv4). The World today, acknowledges the importance and need of transition to IPv6. The need of IPv6 arose mainly because of the exhaustion of IPv4 address blocks. Bangladesh is also entering into the IPv6 transition phase. This paper aims to give an overview of IPv6, IPv6 Deployment and Implementation, Present status of IPv6 implementation in Bangladesh and proposes recommendations for deploying IPv6 in accordance with the present Internet infrastructure in Bangladesh.

P 149: User-Authentication Approach for Data Security Between Smartphone and Cloud

M. A. Hasan	CUET
M. O. Rahman	CUET
M. A. Uddin	CUET

Cloud computing architecture provides a proper management to share distributed resources and services throughout the world via computer network. This architecture offers three main features, e.g. SaaS, PaaS and IaaS. And today's Smartphones are more compatible with this architecture, especially with IaaS because of its small storage capacity. Smartphones have become almost computer and these can be viewed as a miniature of personal computer. Since cloud computing share distributed data via network in the open environment so, there may occur security problems. To address this problem, this paper has proposed a new data security approach for Smartphone in cloud computing architecture, which ensures secured communication system and hiding information from others. Security level is maintained using the Global Positioning System (GPS) and network provider which ensures strong user-authentication to secure our cloud.

P 150: A Binary Code Lock System

W. R. Abrar	CU
M. F. Kader	CU
M. Chowdhury	CUET

The proposed system is intended to protect electrical appliances with a simple lock system. As for passcode, a 4 bit binary number is used. To input the pass code, four two state switches are utilized. The pass code is stored in a memory. It can be changed after unlocking the lock system using valid pass code. We use D type flip-flops as memory. The comparator unit compares input codes with the stored code to generate the signal required to open the lock. The system is hardware implemented and has the versatility in application.

P 172: Design an efficient MAC Protocol for Cognitive Radio Systems using Game Theory

D. K. Saha	CUET
Asaduzzaman	CUET
M. O. Rahman	CUET

In this paper we consider power distribution between cognitive users which is the most crucial problems in cognitive radio systems. To ensure perfect distribution game theory is applied. Conflicting users who are trying to access the same channel are detected first. Targeted signal to noise ratio (SNR) is achieved by iterative fashion which determines the signal strength. Power is calculated for each user by convergence theorem. Power is distributed in such way that it increases system utility. At last Nash equilibrium point is formulated for ensuring the avoidance of selfish behavior of cognitive users. Experimental results show that the proposed strategy achieves the ability to reduce the power consumption in order to increase the system utility.

P 179: An Efficient Approach to Save Cluster Head Energy of Sensor Network

A. Karmaker BUET
M. M. Hasan BUET

Wireless sensor network is an emerging technology in wireless network. Improving the lifetime of sensor node is the major issue for designing the sensor network. For these reason the main concern of researchers is how to utilize the medium in a power effective manner. With advance of these, various MAC protocols are introduced and lots of recent works are going on. But, most of these MAC fail to maintain the effective tradeoff between power consumption and latency. LEACH protocol is one of them which are based on clustering techniques. In this paper, we propose a method for intra cluster communication techniques that decrease power consume by cluster head node also latency and compare with traditional LEACH protocol.

P 180: Software Defined Radio for PC to PC Data Communication

A. H. Rima CUET
H. Hyder CUET
M. A. Ullah CUET

A combination of hardware and software technologies where some or all of the radios operating functions are implemented through adjustable software or firmware operating on programmable processing technologies is defined as Software Defined Radio (SDR). The basic concept of SDR is that the radio functions are configured by software and number of areas can be covered by using same platform. As software is used, configurations can be changed according to various radio functions which are not captured by classic radio. The tasks to be performed included the channels configuration, the management of the data transfer between two PC, the baseband data modulation and demodulation, and the data organization into packets solely by software. For the physical layer of this system Orthogonal Frequency-Division Multiplexing (OFDM) is chosen as the transmission multiplexing method. This choice has been made because of the advantages that OFDM has better channel capacity and provides larger data rates. In this paper, simple IR circuit is used as SDR platform. This IR circuit work as a transceiver which can transmit text, number or image from one PC to another PC. In order to verify the proper functionality of the communication scheme, the received data streams are further analyzed with the use of MATLAB.

P 184: Linear Polarization Switchable Patch Array Antenna using Magic-T Bias Circuit and Orthogonal Feed

M. A. Hossain CUET
P. Chowdhury CUET
Q. D. Hossain CUET
M. A. Rahman CUET
R. S. Goopta CIU

In this paper, a linear polarization switchable patch array antenna is proposed. The orthogonal feed circuit and magic-T circuit is introduced to realize the proposed array antenna. The advantage of the magic-T circuit is the excellent isolation between the RF signal and the switching bias signal. The microwave integration technology is effectively employed to realize proposed linear polarization switchable array antenna. The proposed array antenna consists of four patch elements and 16 PIN diodes. In order to realize the ±45º polarization switching, four switching diodes are integrated with each patch elements. Using the ON/OFF condition of the diodes, the polarization axis can be easily switched to ±45º. The array antenna is realized in very simple and compact structure as all the antenna elements, feeding circuit and bias circuit are arranged on both sides of a dielectric substrate. The ability of the proposed array antenna to switch the polarization axis at ±45º at 10 GHz (X band) is confirmed by the experimental investigation.

Technical Session 2B 03:30~05:30	**MM Lab, Dept. of CSE**
Chair: Prof. Dr. Md. Shahat Hossain	CU
Cochair: Dr. Quazi Delwar Hossian	CUET

P 166: Development of Asynchronous Replication Model for Heterogeneous Environment

M. A. Kashem DUET
M. Naderuzzaman DUET
M. H. Rahman DUET

Replication is a very useful technique in distributed systems, grid community and clustering systems. Now a day a lot of algorithm development has been focused for safe replication services. Replication can improve performance, reliability, portability of entire database. In this paper, a persistent layer has been developed and proposed which supports heterogeneous system. The persistent layer work on asynchronous model, hence it is known as asynchronous replication. The implementation of this algorithm has built on Java based technology thus it is very easy to deploy any OS without hassle and configurable files makes the whole system easy to maintain and cost effective. The experimental servers (both main server and replication server) used both windows and Linux OS's. Finally some experiments have been carried out by different data taken from references. The results show that, the proposed model outperforms all available replication models.

P 169: A Sequence Alignment Algorithm and Tools for Molecular Replacement

M. I. Khan CUET
M. K. Rashikh CUET

This paper describes a new genetic alignment algorithm and software tool for sequences that can be used for determination of deletions and substitutions. Sequence alignment is one of the most active ongoing research problems in the field of computational molecular biology. Sequence alignment is important because it allows scientists to analyze protein strands (such as DNA and RNA) and determine where there are overlaps. This overlaps can show commonalities in evolution and they also allow scientists to better prepare vaccines against viruses, which are made of protein strands. The algorithm provides several solutions out of which the best one can be chosen on the basis of minimization of gaps or other considerations. The algorithm does not use similarity tables and it performs aspects of both global and local alignment. It is also compared with other sequence alignment algorithms.

P 170: Performance Evaluation of Warshall Algorithm and Dynamic Programming for Markov Chain in Local Sequence Alignment

M. I. Khan CUET
M. S. Kamal CUET

Markov Chain is very effective in prediction basically in long data set. In DNA sequencing it is always very important to find the existence of certain nucleotides based on the previous history of the data set. We imposed the Chapman Kolmogorove equation to accomplish the task of Markov Chain. Chapman Kolmogorove equation is the key to help the address the proper places of the DNA chain and this is very powerful tools in mathematics as well as in any other prediction based research. It incorporates the score of DNA sequences calculated by various techniques. Our research utilize the fundamentals of Warshall Algorithm (WA) and Dynamic Programming (DP) to measures the score of DNA segments. The outcomes of the experiment are that Warshall Algorithm is good for small DNA sequences on the other hand Dynamic Programming are good for long DNA sequences. On the top of above findings, it is very important to measure the risk factors of local sequencing during the matching of local sequence alignments whatever the length.

P 171: A New Approach of Disk Scheduling Algorithm

M. A. Kashem DUET
S. Saha DUET
M. Naderuzzaman DUET

The operating system is responsible for using resources efficiently. The disk is of course, one of the computer resources. Computer processing speed depends on disk speed. Processor speed and memory capacity is increasing several times faster than disk speed. This disparity suggests that disk I/O performance will become an important bottleneck. Disk performance management is an increasingly important aspect of operating system research and development. Scheduling is a fundamental operating system function. People have improved I/O performance by intelligent scheduling of disk. In this paper we introduce a new disk scheduling algorithm that reduce the number of head movement by proper scheduling therefore it maximizes throughput for modern storage devices.

P 124: e-Government and Its Implementation Challenges in Nepal

S. Shakya TU

e-Government is the use of Information and Communication Technologies to promote more efficient and effective government, and make it more accessible and accountable to the citizens. The e-readiness index of Nepal is found to be low as compared to other countries. There are various challenges for the implementation of e-government in Nepal. These challenges are like low literacy, low per capita income and limited financial resource. In this paper a conceptual framework is suggested for the effective implementation of e-government in Nepal. The conceptual framework can be further validated in the real life situation.

P 130: An Integrated Online Courseware Design Approach

G. M. M. Bashir PSTU
A. S. M. L. Hoque BUET
S. Majumder PSTU
B. Bepary PSTU
K. Rani PSTU

E-learning helps people to acquire knowledge by using technology at any time and place. Courseware is an efficient way which covers most of the requirements of e-learning. Online courseware has become a demanding form of e-learning in present days. It helps to convert the conventional process of learning and teaching towards e-learning. This paper attempts to give an overview of some online renowned courseware. The authors make a comparison among the courseware and summarize them in the form of tree and table for easy demonstration of those. After studying the similarities and dissimilarities among various contents provided by those courseware, the authors propose a design approach for an integrated global online courseware which will combine all the facilities of the analyzed courseware into a single one. In order to accomplish the desired integration, authors have proposed a methodology with proper demonstration by Venn diagram. If the proposed global online courseware can be developed, then both the learners and instructors will be benefitted. Because of integrating all relevant contents to relevant courses, it will be more beneficial for the third world students and teachers also. Even though integrating all courseware contents depends on the permission of the concerned authority of the courseware, nevertheless global courseware pretends to be the necessity of modern era, especially for the self learners.

P 146: Intelligent Decision System for Evaluation of Job Offers

T. Mahmud	CU
J. Sikder	CU

The word 'Job' term as a regular activity performed in exchange for payment is considered as one of the most important activities for many families worldwide .Evaluation is necessary when more than one opportunity come to an individual personality. Then it requires the job offer evaluation. To fulfill their desired goal, it is the' evaluation' which assesses them well. This involves many factors to be measured and evaluated. These factors are expressed both in objective and subjective ways where as a hierarchical relationship exists among the factors. In addition, it is difficult to measure qualitative factors in a quantitative way, resulting incompleteness in data and hence, uncertainty. Besides it is essential to address the subject of uncertainty by using apt methodology; otherwise, the decision to choose a job will become inapt. Therefore, this paper demonstrates the application of a novel method named Evidential Reasoning (ER) based intelligent decision system (IDS), which is capable of addressing the uncertainty of multi-criterion problem, where there exist factors of both subjective and objective nature. The ER method handles uncertainties by using a belief structure is aggregating degrees of belief from lower level factors to higher level factors.

P 176: Blended Learning Approach for Engineering Education-An Improvement Phase of Traditional Learning

G. M. M. Bashir	PSTU
A. S. M. L. Hoque	BUET
M. J. Hossain	PSTU
M. M. Rahman	PSTU
B. C. Debnath	PSTU

Engineering education is so complex to understand when teaching by a teacher face-to-face (F2F) with the students in our academic education system. This traditional teaching approach cannot make interesting the lecture at classroom and student cannot get attention to the class due to the absence of interactive presentation. To overcome the problem of the traditional teaching of engineering education we demonstrate architecture of blended learning (BL) that helps learners to bring their experiences and ideas to the intellectual conversion, the understanding of the other participants is enriched, resulting in active learning. This paper describes various event-based activities, including F2F classrooms, live eLearning, and self-paced learning. We have proposed a systematic way of BL approach with the existing infrastructure of any institutions. Organizations must use BL approaches in their strategies to get the right content in the right format to the right people at the right time. BL combines multiple delivery media that are designed to complement each other and promote learning and application-learned behavior especially in engineering education. A prototype system is developed and tested using different learning scenarios. The system has also been tested by a group of students.

Important Dates

Paper Submission:	**December 15, 2013**
Acceptance Notification:	**December 30, 2013**
Camera-ready Paper:	**January 15, 2014**

Submission Guidelines

- The authors should submit their NCICIT 2013 conference paper with reasonable extension which is not currently under review by other journals or conferences.
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A Novel Approach of Multilevel Feedback Queue Scheduling

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***Abstract* – Multilevel feedback scheduling is a kind of process scheduling mechanism where process doesn't come with any priority. According to the CPU burst needed by the process and the CPU burst remaining the processes are shifted between queues of the feedback scheduler to get completed. In multilevel feedback queue the total architecture is divided into multiple prioritized queues. In each queue the processes get access of CPU for a fixed time period end after this access time the process is shifted to the next low priority queue with remaining CPU burst time. In this paper we propose a new multilevel feedback queue scheduling technique where based on the remaining CPU burst time we either execute the process immediately or shift the process to next queue. In comparison to other types of MLFQs the performance of the proposed scheduling technique is better and practical according to the consequence.**

***Keywords*— CPU burst time, turnaround time, feedback analysis, starvation, response time**

I. INTRODUCTION

Scheduler is the component of the kernel that selects which process to run next [1]. The scheduler (or process scheduler, as it is sometimes called) can be viewed as the code that divides the finite resource of processor time between the runnable processes on a system. In Multilevel Feedback Queue [2][3] processes are scheduled according to their remaining CPU burst time and they are shifted down from queue to queue as they have some remaining CPU burst time [4]. Every queue has unique time slice that gradually increases from upper level queue to lower level queue. So the CPU intensive jobs go down from upper queues to lower queues gradually for getting completed. Thus lower priority queues are filled with CPU intensive jobs and as a result these processes start to starve for getting CPU attention. So then it will follow first come first serve scheduling among these jobs. Here interactive job means the jobs which go for input and output operations frequently compare to the jobs which are more focused on getting CPU cycles which are considered as CPU intensive jobs. The drawback is found that when a process finish its time slice [4] if a little burst time remains then it is shifted to next queue and it must wait for the CPU until all processes before this in the queue is finished. As a result a severe slowdown in the scheduling and increase the response time and turn around time of the remaining starved processes. Here severe slowdown means that waiting time of the processes are getting increased while residing in the lowest queue. The architecture of MLFQ in this paper we propose, that dynamically reduce the response time of the processes that go down to the lowest queue and as a whole decrease the turn around time of whole scheduling.

II. RELATED WORK

Different approaches are used to increase the performance of MLFQ scheduling [5] in different ways. In paper [6], Recurrent Neural Network has been utilized to optimize the number of queues and quantum of each queue of MLFQ scheduler to decrease response time of processes and increase the performance of scheduling. Here this proposed neural network takes inputs of quantum of queues and average response time and getting the required inputs it takes the responsibility of finding relation between the specified quantum changes with average response time. It can find the quantum of a specified queue with the help of optimized quantum of lower queues. Thus, this network fixed changes and specify new quantum which overall optimize the scheduling time. In the paper [6] smoothed competitive analysis is applied to multilevel feedback algorithm. Smoothed analysis is basically mixture of average case and worst case analysis to explain the success of algorithms. This paper analyses the performance of multilevel feedback scheduling in terms of time complexity. Any performance enhancing approach can use this approach for performance analysis in terms of time complexity. In another paper [7], multilevel feedback queue scheduling algorithm is implemented in Linux 2.6 kernel and new Linux2.6 scheduler performance compared with the proposed approach. It describes two algorithms elaborately and then for different load of job, which are running in background, this scheduler is applied for calculating the average response time. And to maintain simplicity inverse relationship is maintained between priority of processes and time slice length. This paper is a real guideline for designer of cognitive scheduling systems. Now to analyze the performance enhancement, the proposed approach is implemented and simulated in Condor [8] which provides high throughput computing environment , that handles job queue mechanism, scheduling policy, priority scheme, resource monitoring and resource management, based upon the policy fixed for execution Condor executes the job when user submits the job. For compute intensive jobs Condor is very useful. Basically, Condor pool is composed of more than one machine those are under one main machine called central manager [9]. Pool is a collection of machine and jobs. Job submitted in Condor is basically executed depending upon the Class Ad [9] of machine where it is being submitted and when Class Ad of machine matches then that job is executed in that machine. This is called matchmaking [9]. Now for simulation based approach Condor is used for simulation of the comparison based analysis of two scheduling policies which are Round Robin Opportunistic policy and Multi-level Queue Opportunistic policy in the paper [10]. By effective study of this paper reveals that, for simulation based performance analysis of different type of scheduling policies, Condor is one of the effective way that provides high throughput computing environment. Condor not only executes job on standalone machine it also migrates jobs when more than one computer system is joined in the cluster and it finds idle machine to share the workload for better system throughput, basically in this paper [11] it is elaborated that in Condor when migrates job it checkpoints the current job and sent it to the idle remote machine which belong to it’s cluster, for execution. Thus, for better computing Condor is one of the effective approach. In context to this, a leverage quantity is calculated which is the ratio of remote resources utilized and the sending machines utilized resources to support job migration, check pointing and supporting system calls. This leverage quantity is really a matter of issue of any high throughput computing environment for checking environment efficiency. Apart from that check pointing and job migration for data intensive application can create heavy traffic in the total grid, now Condor can handle this network traffic and manages network resources efficiently which is elaborated in the paper [6]. So, Condor not only handles jobs efficiently but also manages network for speedup networking essentials.

III. LITERATURE SURVEY

Different Available Scheduling Algorithms and their Characteristics

A. Priority scheduling

Priority scheduling [4] algorithms assign each process a priority level, which is represented by a number. Some systems consider low numbers to have high priorities and others designate the high numbers to indicate a higher priority. These priorities can be assigned internally (by the operating system) or externally (by the user). Considering some sort of measurable quantity, such as how many resources each process requires, usually sets internal priority levels. External priority levels reflect a user-specified order of importance for each process. The process with the highest priority is initially given control of the CPU. If the algorithm is preemptive, then the currently running process will be preempted once a process in the ready queue has a higher priority. However, if the algorithm is non-preemptive, then once the process begins running all other processes will have to wait until it releases the CPU, regardless of what their priority levels may be. The major concern with priority scheduling is the concept of starvation. This occurs when a process has such a low priority that it is never given control of the CPU. A solution to this is aging, which gradually increases the priority of a process over time.

B. Round-Robin scheduling

The Round-Robin scheduling [4] algorithm (RR) defines a time quantum, which is the amount of time each process in its queue is allocated the CPU. This algorithm is preemptive and fills its queue using the FCFS method. After a process has exceeded its time limit, it is moved to the end of the queue and the next process in the queue is given control of the CPU. Setting a correct time quantum is critical to the performance of the RR algorithm. If the time quantum is too small, then few processes will complete the first time they are run and much time will be spent performing context switches. Alternatively, if the time quantum is too large, then most processes will finish and the CPU will sit idle for the remainder of the quantum. In both of these cases, the CPU is not being used efficiently. Generally, it is desirable for around 80 percent of the processes to complete before the time quantum is reached.

C. First-Come, First-Served

First-Come, First-Served (FCFS) [4] is the most basic scheduling algorithm. It simply allocates the CPU to the process that requested it first. All other processes that request the CPU are placed in the ready queue and are served in the order in which they arrive. This algorithm is non-preemptive; once a process is given control of the CPU it will keep it until the process releases it. A process will release the CPU when it has either finished executing, or if it needs to wait for an I/O event to occur before it can resume its execution.

D. Shortest-Job-First

Shortest-Job-First (SJF) [4] examines the amount of time each process is expected to take during its next CPU burst. It uses this information to form a queue, and then gives control of the CPU to the process that is expected to have to shortest next CPU burst. In theory, this algorithm is very efficient, because it minimizes the average waiting time for each process. However, since the next CPU burst time can be very difficult to accurately determine, this algorithm does not always produce the intended results

E. Multilevel queue scheduling:

A multilevel queue [4] scheduling algorithm separates the ready queue into a number of different queues. Each of these queues uses their own scheduling algorithm. For instance, there may be separate queues created for CPU-bound and I/O-bound processes. The multilevel queue algorithm is responsible for allocating the CPU to each of these different queues. Any scheduling algorithm can be used to achieve this goal; however, fixed-priority preemptive scheduling is commonly used. If the CPU-bound queue were assigned the higher priority, then the CPU would not be given to the I/O-bound queue until all of the processes in the CPU bound queue had completed. A multilevel queue is useful when the processes can be easily grouped by importance.

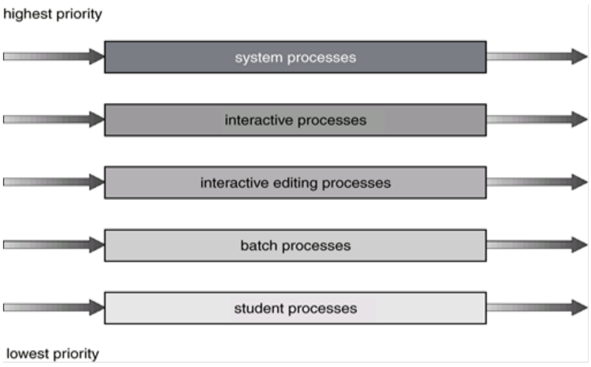


Figure 1: Multilevel queue scheduling

F. Multilevel feedback queue scheduling:

In a multilevel queue [4] scheduling processes are permanently assigned to a queue on entry the system. Processes do not move between queues. This setup has the advantages of low scheduling overhead, but the disadvantages of being inflexible. Multilevel feedback queue scheduling, however, allows a process to move between queues. The idea is to separate processes with different CPU-burst characteristics. If a process uses too much CPU time, it will be moved to a lower-priority queue. This scheme leaves I/O bound and interactive processes in the higher-priority queues. Similarly, a process that waits too long in a lower-priority queue may be moved to a higher-priority queue. This form of aging prevents starvation.

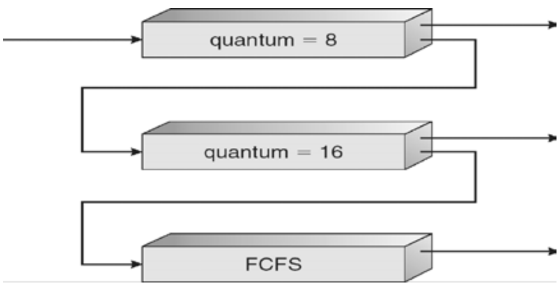


Figure 2: Multilevel feedback queues

A multilevel-feedback-queue scheduler is defined by the following parameters:

- The number of queues.
- The scheduling algorithms for each queue.
- The method used to determine when to promote a process.
- The method used to determine when to demote a process.
- The method used to determine which queue a process will enter when that process needs service.

IV. PROPOSED APPROACH

In this new scheduling approach we use different queues with a fixed time slice for each queue. When a process comes to the ready state it is inserted in first queue with small slice time. When preceding process of the process finish their time slice then CPU is assigned to this process. Then it accesses the CPU for given time period. After the time slice it calculates the remaining burst time. If the remaining burst time greater than half of the time slice of the current queue then the process is inserted to next queue with a larger time slice and lower priority. But if the remaining burst time is smaller than half of the time slice of the current queue then the CPU is assigned to the process and it finish its execution immediately without waiting. Therefore processes with small remaining burst time do not need to wait for other processes in next lower priority queue. The algorithm, control flow diagram and result elaborate it fully.

V. PROPOSED ARCHITECTURE

In proposed architecture there are three queues for example .The proposed architecture is drawn in the Figure 3. The processes, which will be scheduled, will come to the queue 1 with 8 quantum and it will go downwards to the lower priority queues till get finished. The ready will supply the processes information while the proposed architecture schedules the processes and the results of scheduling will be given as output. Now one major common issue of this architecture is that the number queues are not constant but the time slice of each queue is increased from upper to lower.

The architecture supports the scheduling policy of feedback analysis where processes do not have any fixed priority in the beginning according to the CPU burst and the time slice of each queue they are scheduled the proposed algorithm elaborates the scheduling mechanism in details. But the priority of queues decreases form smaller slice time queue to larger slice time queue. When the first queue will be empty then the access is handled to the second queue and similar for other queues.

From architecture it is clear that all process get the CPU in Round-Robin approach in each queue. The difference occurs when the remaining burst time of current process smaller than half of quantum time. In this case the process gets the CPU access immediately.

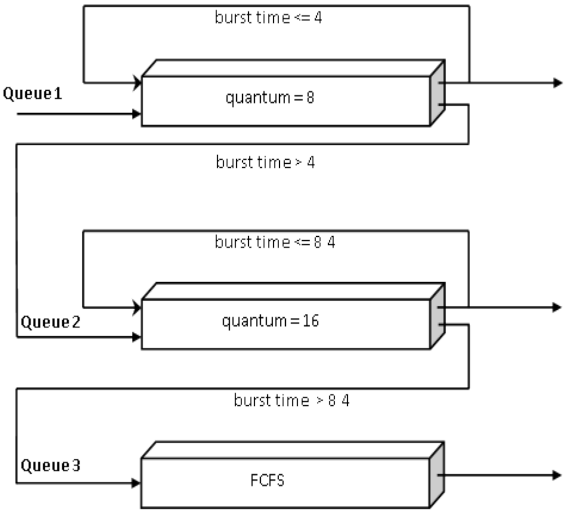


Figure 3: Proposed architecture

VI. PROPOSED ALGORITHM

In this proposed algorithm three queues are used, for quantum time 8 Q8, for quantum time 16 Q16 and for FCFS we used QFCFS. All processes are inserted first in a two dimensional array P[][]. Each process is handled by an identifier i.

The algorithm is listed bellow

Algorithm: New Multilevel Feedback Queue Scheduling()

//P[][] is the array of processes with burst time

//Q8 and Q16 are queues with slice time 8 and slice time 16

//QFCFS is the last queue with low priority

1. Enter number N of processes
2. For i = 1 to N do
3. Enter burst time of process P[i]
4. While there exist any process in Q8, Q16 or QFCFS do
5. If any process exist in Q8 then
6. P[i] access CPU
7. If burst time of P[i] less than 5 then
8. P[i] complete execution immediately
9. Else P[i] is inserted into Q16
10. If any process exist in Q16 then
11. P[i] access CPU
12. If burst time of P[i] less than 9 then
13. P[i] complete execution immediately
14. Else P[i] is inserted into QFCFS
15. If any process exist in QFCFS then
16. P[i] completes execution
17. Show result.

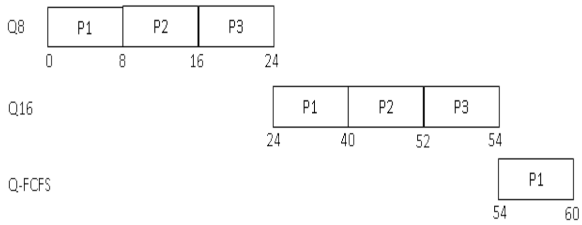
VII. GANTT CHART

Following example elaborates the performance enhancing scenario with suitable Gantt chart and discussions. Suppose, MLFQ scheduler has three queues having time slice suppose 8, 16 and first come first serve. This time slice assumption is taken to show the increasing order of time slice and power MLFQ property. In this example we consider three processes these are P1 with burst time 30, P2 with burst time 20 and P3 with burst time 10. First we consider Multilevel Feedback Queue scheduling and then new scheduling approach.

Process Identifier	Burst Time
P1	30
P2	20
P3	10

Multilevel Feedback Queue Scheduling:

Initially P1=30 P2=20 P3=10



Waiting time of P1 = 0 + (24 – 8) + (54 – 40) = 30

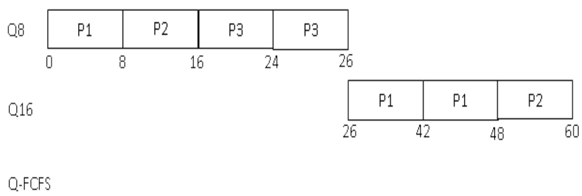
P2 = 8 + (40 – 16) = 32

P3 = 16 + (52 – 24) = 44

Total waiting time = 30 + 32 + 44 = 106

New Multilevel Feedback Queue Scheduling:

Initially P1=30 P2=20 P3=10



Waiting time of P1 = 0 + (26 – 8) = 18

P2 = 8 + (48 – 16) = 40

P3 = 16

Total waiting time = 18 + 40 + 16 = 74

Percentage of reduced waiting time = 74*100/106 = 71.15

VIII. CONCLUSION

In our article it is clearly proved that the new scheduling technique reduces the waiting time of each interactive process and at the same time it also reduces total waiting time of the system. Here a process with remaining burst time less than half of the quantum time does not need to wait for CPU. If the remaining burst time is larger than half of quantum time only this process is shifted to next queue. So it could be said the new approach gives a better efficiency of the system and CPU. Our approach extends the performance of feedback scheduling algorithm by minimizing the total waiting time of the system by around 71%.

REFERENCES

[1] M. V. Panduranga Rao and K. C. Shet, "Analysis of New Multi Level Feedback Queue Scheduler for Real Time Kernel", International Journal of Computational Cognition (http://www.ijcc.us), vol.8, no. 3, September 2010.

[2] Hoganson, Kenneth (2009), "Reducing MLFQ Scheduling Starvation with Feedback and Exponential Averaging", Consortium for Computing Sciences in Colleges, Southeastern Conference, Georgia.

[3] Parvar, Mohammad R.E, Parvar, M. E. and Safari, Saeed (2008), "A Starvation Free IMLFQ Scheduling Algorithm Based on Neural Network", International Journal of Computational Intelligence Research ISSN 0973-1873 Vol.4, No.1 pp. 27–36

[4] Operating System Concepts by Abraham Silberschatz, Peter Baer Galvin, Greg Gagne.

[5] Ayan Bhunia, "Enhancing the Performance of Feedback Scheduling" International Journal of Computer Applications (0975 – 8887), Volume 18 No.4, March 2011.

[6] Becchetti, L., Leonardi, S. and Marchetti S.A. (2006), "Average-Case and Smoothed Competitive Analysis of the Multilevel Feedback Algorithm" Mathematics of Operation Research Vol. 31, No. 1, February, pp. 85–108.

[7] Basney, Jim and Livny, Miron (2000), "Managing Network Resources in Condor", 9th IEEE Proceedings of the International Symposium on High Performance Distributed Computing, Washington, DC, USA.

[8] Scheduling: The Multilevel Feedback Queue. http://pages.cs.wisc.edu/~remzi/Classes/537/Fall2009/Notes/cpu-sched-mlfq.pdf

[9] Liu, Chang, Zhao, Zhiwen and Liu, Fang (2009), "An Insight into the Architecture of Condor –A Distributed Scheduler", IEEE, Beijing, China.

[10] Condor Team, University of Winconsin-Madision, "Condor Version 7.2.5 Manual".

[11] Abawajy, Jemal H. (2002), "Job Scheduling Policy for High Throughput Computing Environments", Ninth IEEE International Conferences on Parallel and Distributed Systems, Ottawa, Ontario, Canada.

[12] Litzkow, Micheal J., Linvy, Miron and Mutka, Matt W. (1988), "Condor – A Hunter of Idle Workstations" IEEE, Department of Computer Sciences, University of Wisconsin, Madison.

[13] Wolski, Rich, Nurmi, Daniel and Brevik, John (2007), "An Analysis of Availability Distributions in Condor", IEEE, University of California, Santa Barbara.

e-Government and Its Implementation Challenges in Nepal

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Abstract -E-Government is the use of Information and Communication Technologies to promote more efficient and effective government, and make it more accessible and accountable to the citizens. The e-readiness index of Nepal is found to be low as compared to other countries. There are various challenges for the implementation of e-government in Nepal. These challenges are like low literacy, low per capita income and limited financial resource. In this paper a conceptual framework is suggested for the effective implementation of e-government in Nepal. The conceptual framework can be further validated in the real life situation.

Keywords

e-Government, e-readiness, low literacy, Low per Capita, conceptual framework

1. INTRODUCTION

E-Government is the use of information technology to provide citizen and organizations with more convenient access to government information and services and to provide delivery of public services to citizen, business partners, and those working in the public sector. In other words e-Government is the use of information and communication technologies (ICTs) to improve the activities of public sector organizations. E-Government has very significant role in the development of a nation. It has brought revolution in the governance of the government. According to Economist Intelligence Unit the e-readiness index of Nepal is low. The implementation of e-government is very challenging because of the low level of literacy, low per capita income and insufficient infrastructure for the implementation of e-government in Nepal.

2. E-GOVERNMENT PROJECT IN NEPAL

The government of Nepal with support of the Korea IT Industry Promotion Agency (KIPA) prepared an e-Government Master plan in November 2006[1]. In order to establish the foundation for investment phase of the Master Plan, The Asian Development Bank (ADB) provided a project preparatory technical assistance (PPTA) to the Government of Nepal [2]. According to the PPTA, the implementation plan was initiated and ADB agreed to invest grant fund for Nepal and after that the government of Nepal and Asian Development Bank (ADB) signed the grant agreement of ICT development project dated 23 May 2008[3]. Then after e-Government implementation project starts in Nepal. The Project involves[5] (i) modernizing rural communities, particularly in remote areas, by improving rural connectivity through wireless broadband networks, mobilizing community socioeconomic activities through village network portals, and building telecenters to improve last-mile access to services in remote rural areas; (ii) building a government information and communication technology (ICT) network, which allows government-to-government exchange of data and information and central management of government data and information; (iii) developing various priority e-Government applications; and (iv) developing and implementing human resources development programs.

There have been e-Government initiatives in the country at national level. The program will lead to socioeconomic improvements in remote and rural communities through ICT and improved ICT uptake in the community, business, and government. To this end, the program will (i) make ICT more accessible, affordable, inclusive, sustainable, and useful to remote and rural communities; (ii) make public services more citizen-centered and business-friendly through ICT; (iii) improve accessibility, efficiency, and transparency in government service delivery through ICT; and (iv) enhance ICT business and industry. The priority project is as follows [4]:

Rural e-Community

The Project will modernize rural communities, particularly in remote areas, by improving rural connectivity through wireless broadband networks in districts, by mobilizing community socioeconomic activities through a village network portal that will allow villagers to share their social capital, and by building tele centers to improve last-mile access to services in remote rural areas.

Government Network

The Project will build a government ICT network, which allows government-to-government exchange of data and information, and central management of government data and information, with suitable protection and efficient backup and recovery provisions. The government ICT network component will comprise (i) the establishment of government information and data center, and (ii) the development of government groupware.

E-*Government Applications*

The Project will develop various priority e-government applications that (i) are priority applications; (ii) are in areas requiring few legislative changes; (iii) provide easily identifiable benefits; (iv) respond to owners' keenness to initiate change and own the application; (v) are relevant to the immediate needs of the community; (vi) hasten the achievement of MDGs; (vii) answer actual or latent demand, cost, and organizational capability; or (viii) are likely to have quick success, with significant dividends to the community.

Human Resources Development for e-Governance

The lack of adequate knowledge and skills in ICT and e-Governance in the public sector has given rise to (i) poor-quality or nonexistent services; (ii) persistent corruption due to lack of transparency; (iii) unnecessary duplication and inefficiency in government work; (iv) flawed decisions because of a lack of information and analysis; (v) coordination difficulties and resistance to innovation; and (vi) poor use of financial, human, physical, and technical resources. To address this capacity problem, the Project will (i) build awareness, knowledge, and skills in ICT governance among key stakeholders to help improve efficiency in the delivery of e-services to the community; (ii) establish computer laboratories for the capacity development of institutions promoting ICT human resources development, and strengthen networking between training institutions and support for functional linkages; (iii) revise the training curriculum and develop new curricula for public training institutions, to improve the quality of the training curriculum and materials in ICT governance and enhance the cost-effectiveness of training programs by developing high-quality training materials; (iv) share knowledge and experiences to promote ICT governance and applications through exchange and fellowship programs designed to recognize the contributions and commitment of exemplary civil servants and NGO leaders to the effective and efficient adoption of ICT governance applications; and (v) support the development of ICT governance courses, a new curriculum for ICT governance, and new teaching materials in ICT governance, in association with universities and research institutes.

3. IMPLEMENTATION ARRANGEMENTS

The Office of the Prime Minister and Council of Ministers (OPMCM) [4] will be responsible for overall project implementation and coordination as the Executing Agency. A project management unit (PMU) will be established in the OPMCM, within one month of effectiveness. It will be headed by a chief project director (secretary), who will be supported by two project directors, one for administration (joint secretary) and another for technical aspects (NITC executive director), and a treasurer, as well as project management consultants, who will provide technical inputs for the appropriate supervision of project implementation. The chief project director will have overall responsibility for project management. Each component and subcomponent of the Project will be assigned an implementing agency as follows: (i) for the rural e-community component, the Ministry of Information and Communications (MOIC); (ii) for the government network component, the Ministry of Environment, Science and Technology (MOEST); (iii) for the enterprise architecture subcomponent, HLCIT; (iv) for the national identification system subcomponent, the Ministry of Home Affairs; (v) for the e-governance in the Public Service Commission subcomponent, the Public Service Commission; (vi) for the land records management subcomponent, the Ministry of Land Reform and Management; (vii) for the vehicle registration and driver's licenses subcomponent, the Ministry of Labor and Transport Management; and (viii) for the human resources development component, the Ministry of General Administration.

Project implementation units (PIUs) will be named by the various Implementing Agencies for their respective components and subcomponents, as follows: NITC, for the government network component and the government representation portal subcomponent; the Department of Land Reforms and Management, for the land records management subcomponent; and the Department of Transport Management, for the vehicle registration and driver's licenses subcomponent. For other components and subcomponents, the Implementing Agency will establish a PIU headed by a project manager. The PIUs will be established within one month of effectiveness and will (i) coordinate with the contractors and monitor the day-to-day progress of project management and implementation, (ii) prepare fund withdrawal applications and submit them to the PMU under the OPMCM for endorsement, (iii) prepare monthly project progress reports and submit them to the PMU under OPMCM, (iv) take charge of procurement for their respective areas of assignment, and (v) keep the project accounts and prepare the annual reports.

Overall project implementation will be supervised by a project steering committee (PSC). The PSC will be established within one month of effectiveness, will have the chief secretary of OPMCM as chair, and will comprise the project director (vice-chair), the secretaries of MOIC and MOEST, the member-secretary of HLCIT, and the executive director of NITC (committee secretary). The PSC will meet regularly at least twice a year, or more often if necessary, to set the direction for overall project implementation, guide the PMU on specific project implementation issues, and resolve any dispute between the PMU and the project implementation agencies. But there is no just type of framework to effective implementation of e-Government in Nepal.

4. NEPAL'S POSITION ON E-READINESS

e-Readiness is the ability to use Information and Communication Technologies (ICT) to develop one's economy and to foster one's welfare. Each year, the Economist Intelligence Unit produces a ranking of e-readiness across countries, based on six pillars of e-readiness: connectivity & technology infrastructure, business environment, social & cultural environment, legal environment, government policy & vision and consumer & business adoption. Nepal is at 153 positions with e-readiness in 2012[5].

	e-Gov. Development Index		World e-Gov. development ranking	
Country	*2012*	*2010*	*2012*	**2010**
Maldives	*0.4994*	*0.4392*	*95*	92
Iran	*0.4876*	*0.4234*	*100*	102
Sri Lanka	*0.4357*	*0.0.3995*	*115*	111
India	*0.3829*	*0.3567*	*125*	119
Bangladesh	*0.2991*	*0.3028*	*150*	134
Bhutan	*0.2942*	*0.2598*	*152*	152
Pakistan	*0.2823*	*0.2755*	*156*	146
Nepal	*0.2664*	*0.2568*	*164*	153
Afghanistan	*0.1701*	*0.2098*	*184*	168
Sub Region Average	*0.3464*	*0.3248*		
World Average	*0.4882*	*0.4406*		

Table 1: e-Government readiness rankings for Southern Asia

(Source: United Nations e-Gov. Survey 2012, Page 26, Table 117)

5. CHALLENGES FOR IMPLEMENTATION OF E-GOVERNMENT IN NEPAL

Implementation of e- Government has changed the way of living of the people in many countries. However, in Nepal the implementation of e-Government is little difficult because of its developing status. The government agencies find lot of difficulties in the smooth implementation of e-government in Nepal because of low literacy, low per capita income, insufficient infrastructure and limited financial resource.

5.1 Low literacy

Literacy is defined as the ability to read and write with understanding in any language. A person who can merely read but cannot write is not classified as literate. Any formal education or minimum educational standard is not necessary to be considered literate. Georgia is the country having 100% literacy rate and at 1st rank in the list of literacy (Table: 2). Literacy level of Nepal is 48.6% and ranked 162 in the list and Mali is having literacy level 24% , is the lowest level in literacy level (176).

Table 2 : Literacy rate of select countries.

S.N.	Country	Literacy Rate	Rank
1	Georgia	100.0	1
2	Cuba	99.8	2
3	Russia	99.4	12
4	Australia	99.0	18
5	Canada	99.0	18
6	Germany	99.0	18
7	Japan	99.0	18
8	Switzerland	99.0	18
9	United Kingdom	99.0	18
10	United States	99.0	18
11	China	99.9	86
12	Sri Lanka	90.7	87
13	Swaziland	79.6	124
14	India	61.0	147
15	Pakistan	49.9	160
16	Nepal	48.6	162
17	Mali	24.0	176

(Source: http://en.wikipedia.org/wiki/List_of_countries_by_literacy_rate)

5.2 Low per capita income

Per capita income means how much each individual receives, in monetary terms, of the yearly income generated in the country. This is what each citizen is to receive if the yearly national income is divided equally among everyone. Per capita income is usually reported in units of currency per year. Globally Bermuda has the highest per capita income followed by Luxembourg. India is at 160th rank in with 530 US$ average per capita income (Table 3).

Table 3: National Average Per Capita Income of select countries

S.N	Country	Per Capita Income in USD	Ranking
1	Bermuda	N/A	1
2	Luxembourg	43,940	2
3	Switzerland	39,880	4
4	United States	37,610	5
5	Japan	34,510	7
6	United Kingdom	28,350	12
7	Germany	25,250	22
8	Canada	23,930	24
9	Australia	21,650	27
10	Swaziland	1,350	127
11	China	1,100	133
12	Sri Lanka	930	140
13	India	530	160
14	Pakistan	470	166
15	Bangladesh	400	174
16	Nepal	240	192
17	Ethiopia	90	208

(Source: http://www.success-and-culture.net/articles/percapitaincome.shtml)

5.3 Limited financial Resource

The Gross Domestic Product (GDP) is one of the measures of national income and output for a given country's economy. GDP is defined as the total market value of all final goods and services produced within the country in a given period of time (usually a calendar year). GDP of a country is the measure of its financial strength. United States is having the highest GDP in the world followed by Japan, Germany, China, United Kingdom (Table 4) and India with GDP 1,237,000 million US $ is at rank 12th (Table 4). Nepal with GDP 12,640 million US $ is at rank 121st (Table 4).

Table 4 : GDP of select countries in 2008 given by the CIA World Fact book

S.N	Country	GDP(Million USD)	Rank
1	United States	14,330,000	1
2	Japan	4,844,000	2
3	China	4,222,000	3
4	Germany	3,818,000	4
5	France	2,978,000	5
6	United Kingdom	2,787,000	6
7	Russia	1,757,000	8
8	Brazil	1,665,000	10
9	Canada	1,564,000	11
10	India	1,237,000	12
11	Australia	1,069,000	14
12	Switzerland	492,600	22
13	Pakistan	160,900	49
14	Sri Lanka	42,160	78
15	Nepal	12,640	121
16	Kiribati	71	189

(Source: http://en.wikipedia.org/wiki/List_of_countries_by_GDP_(nominal))

6. A STRATEGIC FRAMEWORK FOR IMPLEMENTATION OF E-GOVERNMENT

According to study of e-readiness in Nepal and found the implementation challenges of e-Government in Nepal. A conceptual framework is suggested for the effective implementation of e-government in Nepal. The five stages of the conceptual framework is suggested as follows:

Vision and Action plan for e-government implementation

In the first stage the vision and action plan for the effective implementation of e-Government has to be determined. In this level it is planned that to what extend the e-government can be implemented.

Framework for e-readiness Assessment

In this second stage to fulfill the vision the e- readiness of Nepal should be assessed. In this stage compared with respect to other countries. The e- readiness reveals the position of Nepal with respect to the other countries.

Overcoming challenges of e-Government

In this stage, the assessments process the challenges for effective implementation of e-Government will be exposed.

These challenges are low literacy levels, low per capita income and limited financial resource in Nepal. The challenges should be overcome for the effective implementation of e-government.

Developing the environment for e-Government

In this stage to developed positive environment needs to meet the vision of e- government implementation. This environment is internal and external environment.

Implementation of e-Government

After that the e-government should be implemented. This is the final step of the conceptual framework for e-Government implementation.

A conceptual framework , is shown in figure 1. This is a conceptual framework and can be validated in the real life situation.

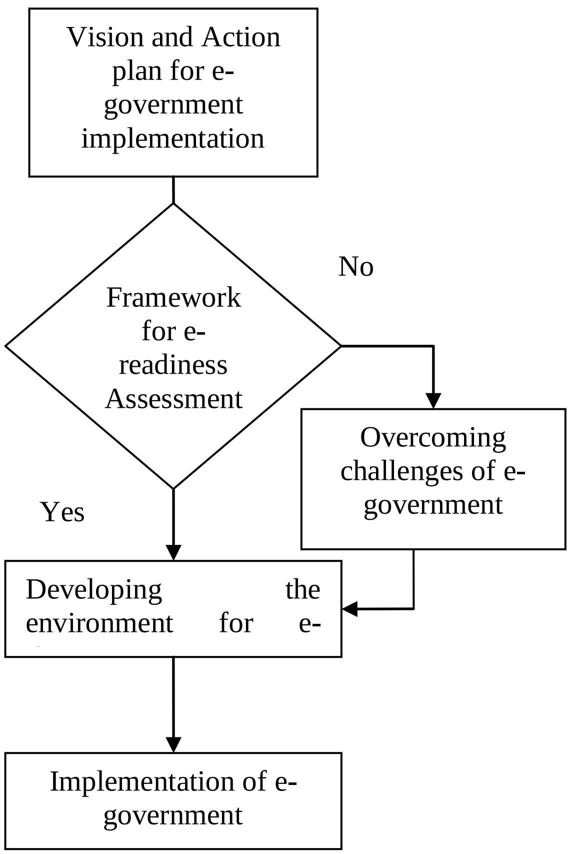

Figure 1: A conceptual framework for e-government implementation in Nepal

7. CONCLUSION

Thus Economist Intelligence Unit depicts that the e-readiness index of Nepal is low. According to this study the implementation of e-Government in Nepal found various challenges. The main challenges are like low literacy, low per capita income, and limited financial resource. A vision and leadership is required to implement the e-Government in Nepal. To meet the vision the challenges in the implementation of e-government should be overcome and needs to be developed environment for the effective implementation of e-government. A conceptual framework is developed for the effective implementation of e-Government in Nepal This conceptual framework and can be further validated in the real life situation.

8. REFERENCES

[1]e-Government Master plan consulting report for Government of Nepal

[2] TA No. 4833-NEP: Preparing the Information and Communications Technology Development Project, a Programmme of Modernisation of Nepal, July 2007.

[3]http://www.adb.org/Documents/Legal-Agreements/NEP/38347/38347-NEP-GRJ.pdf

[4]ADB , Report and recommendation of the President to the Board of Directors, proposed Asian [5]United nations e-Government Survey 2012

[6]http://en.wikipedia.org/wiki/E-readiness accessed during September 2008

[7]http://en.wikipedia.org/wiki/List_of_countries_by_literacy_rate) accessed during September 2008 Development Fund Grant Nepal: ICT development Project, 2007, pp.,7-11, 13

Irregular Total Labellings of Knödel Graphs

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***Abstract*—The total edge irregularity strength *tes(G)* and total vertex irregularity strength *tvs(G)* are invariants analogous to irregular strength *s(G)* of a graph *G* for total labellings. Bača et al. [1] determined the bounds and precise values for some families of graphs concerning these parameters. In this paper, we show the exact values of the total edge irregularity strength $tes(W(3,n)) = \frac{n}{2} + 1$ total vertex irregularity strength and $tvs(W(3,n)) = \lceil \frac{n}{4} \rceil + 1$ for the Knödel Graphs *W (3, n)*.**

Keywords: Irregular total labelling; Knödel Graphs; Total labeling.

1. INTRODUCTION

We consider only finite undirected graphs without loops or multiple edges. Let *G = (V, E)* be a graph with vertex set *V* and edge set *E*.

An edge irregular total *k*-labelling of a graph *G* is a labelling of the vertices and edges with labels *1,..., k* such that for every two different edges their weights are distinct where the weight of an edge is the sum of its label and the labels of its two end vertices. A vertex irregular total *k*-labelling of a graph *G* is a labelling of the vertices and edges with labels *1,..., k* such that for every two different vertices their weights are distinct where the weight of a vertex is the sum of its label and the labels of its incident edges. The minimum *k* for which the graph *G* has an edge irregular total *k*-labelling is called the total edge irregularity strength of the graph *G*, *tes(G)*. Analogously, the minimum *k* for which there exists a vertex irregular total *k*-labelling is called the total vertex irregularity strength of *G*, *tvs(G)*.

The notions of the total edge irregularity strength and total vertex irregularity strength were first introduced by Bača et al. [1]. They may be taken as an extension of the irregularity strength of a graph [2, 4, 6, 8, 9, 12, and 13]. In [1], the authors put forward the lower bounds of *tes(G)* and *tvs(G)* in terms of the maximum degree Δ, minimum degree δ, |E(G)| and |V (G)|, which may be stated as the Theorems 1.1 and 1.2:

Theorem 1.1. $tes(G) \geq \max\left\{\left\lceil \frac{\Delta+1}{2} \right\rceil, \lceil\right.$

Theorem 1.2. $tvs($

Bača et al. [1] then determined the exact values of the total edge irregularity strength for path P_n, star S_n, wheel W_n and friendship graph F_n, and obtained the exact values of the total vertex irregularity strength for star S_n, complete graphs K_n, cycle C_n and prism D_n. The author proved that irregular total labellings of Generalized Petersen graphs *P(n, k)* and Irregular Total Labelling of Möbius Ladder in [10, 11]. We refer the readers for some recent results [3, 5, and 7]. In this paper, we deal with the Knödel Graphs.

The *Knödel graph W (3, n)* has even n ≥ *2* vertices and degree 3. The vertices of *W (3, n)* are the pairs *(i, j)* with *i* = 1, 2 and . For every *j*, , there is an edge between vertex *(1, j)* and every vertex $\left(2, (j + 2^k - 1) mod \left(\frac{n}{2}\right)\right), for\ k = ($.

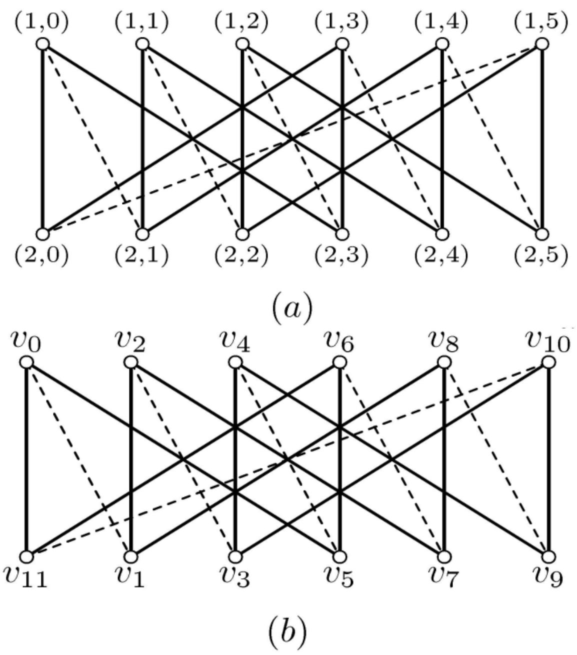
Let v_j represent vertex *(1, j)* and u_j represent vertex *(2, j)*, then we have

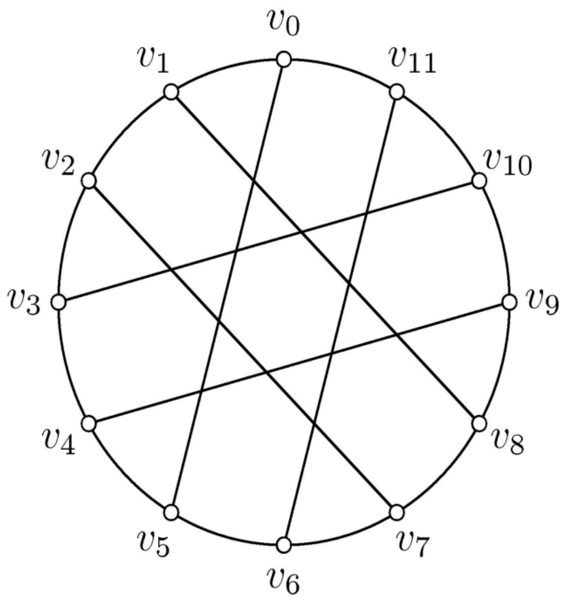
$$V(W_{3,n}) = \left\{v_0, v_1, \cdots, v_{\frac{n}{2}-1}, u_0, u_1, \cdots, u_{\frac{n}{2}-1}\right\},$$

$$E(W_{3,n}) = \bigcup_{t=0}^{n/2-1} \{v_t u_t, v_t u_{t+1}, v_t u_{t+3}, subscripts\ modulo\ \tfrac{n}{2}\}$$

.

In Figure 1.1.1, we show the Knödel graph for *W (3, 12)*.





(c)

Figure 1.1.1. Knödel graph for *W (*3, 12)

2. Main results

2.1 Irregular total labelling of W(3, n)

Theorem 2.2.1. $tes(W$

Proof. Fo we construct the function *f* as follows:

Case 1.

$$
\begin{aligned}
f(v_i) &= \begin{cases} \frac{n}{2}+1, & i \bmod 2 = 0,\\ 1, & i \bmod 2 = 1,\end{cases}\\
f(u_i) &= \begin{cases} 1, & i \bmod 2 = 0,\\ \frac{n}{2}+1, & i \bmod 2 = 1,\end{cases}\\
f(v_iu_i) &= i+1,\\
f(v_iu_{i+1}) &= \frac{n}{2}-i,\\
f(v_iu_{i+3}) &= i+1.
\end{aligned}
$$

Since

$$
wt(v_iu_i) = \begin{cases} (\frac{n}{2}+1)+(i+1)+1 = \frac{n}{2}+i+3, & i \bmod 2 = 0,\\ 1+(i+1)+(\frac{n}{2}+1) = \frac{n}{2}+i+4, & i \bmod 2 = 1,\end{cases}
$$

$$
wt(v_iu_{i+1}) = \begin{cases} (\frac{n}{2}+1)+(\frac{n}{2}-i)+(\frac{n}{2}+1) = \frac{3n}{2}-i+2, & i \bmod 2 = 0,\\ 1+(\frac{n}{2}-i)+1 = \frac{n}{2}-i+2, & i \bmod 2 = 1,\end{cases}
$$

$$
wt(v_iu_{i+3}) = \begin{cases} (\frac{n}{2}+1)+(i+1)+(\frac{n}{2}+1) = n+i+3, & i \bmod 2 = 0,\\ 1+(i+1)+1 = i+3, & i \bmod 2 = 1.\end{cases}
$$

the weights of edges of *W*(3, n) under the labelling *f* constitute the set and the function *f* is a map from

$V(W(3,n)) \cup E(W(3,n))\, into \left\{1, 2, \cdots, \frac{n}{2}+\right.$

for *n mod* 4= 0.

Case 2.

$$
\begin{aligned}
f(v_i) &= \begin{cases} 1, & 0 \le i \le \frac{n}{2}-3 \text{ and } i \bmod 2 = 0,\\ \frac{n}{2}+1, & i = \frac{n}{2}-1,\\ \frac{n}{2}+1, & 1 \le i \le \frac{n}{2}-4 \text{ and } i \bmod 2 = 1,\\ \frac{n}{2}-1, & i = \frac{n}{2}-2,\end{cases}\\
f(u_i) &= \begin{cases} \frac{n}{2}+1, & i \bmod 2 = 0,\\ 1, & i \bmod 2 = 1,\end{cases}\\
f(v_iu_i) &= \begin{cases} i+1, & i \ne \frac{n}{2}-2,\\ \frac{n}{2}+1, & i = \frac{n}{2}-2,\end{cases}\\
f(v_iu_{i+1}) &= \begin{cases} \frac{n}{2}-i, & 0 \le i \le \frac{n}{2}-2,\\ 2, & i = \frac{n}{2}-1,\end{cases}\\
f(v_iu_{i+3}) &= \begin{cases} i+2, & 0 \le i \le \frac{n}{2}-4\\ \frac{n}{2}+1, & i = \frac{n}{2}-3 \text{ or } i = \frac{n}{2}-1\\ 1, & i = \frac{n}{2}-2.\end{cases}
\end{aligned}
$$

$$
wt(v_iu_i) = \begin{cases} 1+(i+1)+(\frac{n}{2}+1) = \frac{n}{2}+i+3, & 0 \le i \le \frac{n}{2}-3 \text{ and } i \bmod 2 = 0,\\ (\frac{n}{2}+1)+(i+1)+(\frac{n}{2}+1) = \frac{3n}{2}+2, & i = \frac{n}{2}-1,\\ (\frac{n}{2}+1)+(i+1)+1 = i+3+\frac{n}{2}, & 1 \le i \le \frac{n}{2}-4 \text{ and } i \bmod 2 = 1,\\ (\frac{n}{2}-1)+(\frac{n}{2}+1)+1 = n+1 & i = \frac{n}{2}-2.\end{cases}
$$

$$
wt(v_iu_{i+1}) = \begin{cases} 1+(\frac{n}{2}-i)+1 = \frac{n}{2}+2-i, & 0 \le i \le \frac{n}{2}-3 \text{ and } i \bmod 2 = 0,\\ (\frac{n}{2}+1)+2+(\frac{n}{2}+1) = n+4, & i = \frac{n}{2}-1,\\ (\frac{n}{2}+1)+(\frac{n}{2}-i)+(\frac{n}{2}+1), & 1 \le i \le \frac{n}{2}-4 \text{ and } i \bmod 2 = 1,\\ (\frac{n}{2}-1)+2+(\frac{n}{2}+1) = n+2, & i = \frac{n}{2}-2.\end{cases}
$$

$$
wt(v_iu_{i+3}) = \begin{cases} 1+(i+2)+1 = i+4, & 0 \le i \le \frac{n}{2}-5 \text{ and } i \bmod 2 = 0,\\ 1+(\frac{n}{2}+1)+(\frac{n}{2}+1) = n+3, & i = \frac{n}{2}-3,\\ (\frac{n}{2}+1)+(\frac{n}{2}+1)+(\frac{n}{2}+1) = \frac{3n}{2}+3, & i = \frac{n}{2}-1,\\ (\frac{n}{2}+1)+(i+2)+(\frac{n}{2}+1) = n+i+4, & 1 \le i \le \frac{n}{2}-4 \text{ and } i \bmod 2 = 1,\\ (\frac{n}{2}-1)+1+1 = \frac{n}{2}+1, & i = \frac{n}{2}-2.\end{cases}
$$

the weights of edges of *W*(3, n) under the labelling *f* constitute the set and the function *f* is a map from

$V(W(3,n)) \cup E(W(3,n))\, into \left\{1, 2, \cdots, \frac{n}{2}+1\right\}$

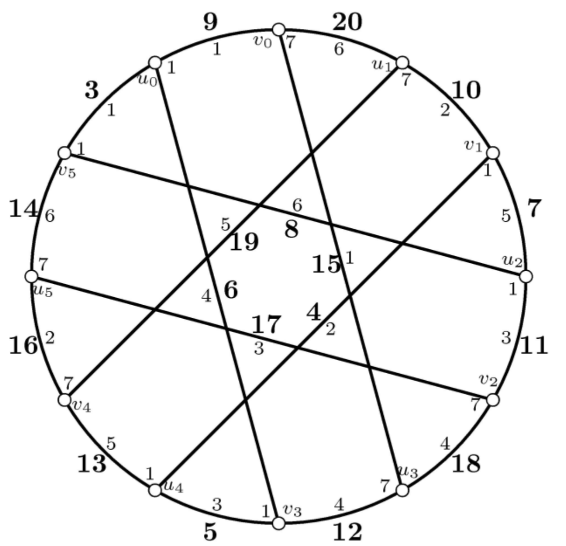
for *n mod* 4= 2.

From case 1 and 2, the total labellings *f* has the required properties of an edge irregular total labelling. We then have $tes(W$.

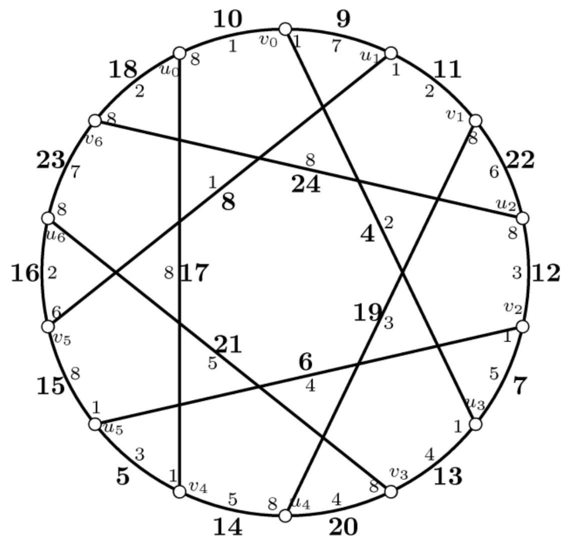
However, by Theorem 1.1, $tes(W(3,n)) \geq \lceil \frac{|E(G)|+2}{3} \rceil = \lceil \frac{\frac{3n}{2}+2}{3} \rceil =$. This concludes the proof.

□

In Figure 2.2.1, we show the edge irregular total labellings for *W* (3, 12) and *W* (3, 14).



W(3,12)



W(3, 14)

Figure 2.2.1. Edge irregular total labelling for *W* (3, 12) and *W* (3, 14)

Theorem 2.2.2. $tvs(W(3,n$

Proof. Fo we construct the function *f* as follows:

Case 1.

$$
\begin{aligned}
f(v_iu_i) &= \lfloor \tfrac{i}{2} \rfloor + 1, \\
f(v_iu_{i+1}) &= \begin{cases} 1, & i \bmod 2 = 0, \\ \frac{n}{4}+1, & i \bmod 2 = 1, \end{cases} \\
f(v_iu_{i+3}) &= \begin{cases} 1, & i \bmod 2 = 0, \\ \frac{n}{4}+1, & i \bmod 2 = 1, \end{cases} \\
f(v_i) &= \begin{cases} 1, & i \bmod 2 = 0, \\ \frac{n}{4}+1, & i \bmod 2 = 1, \end{cases} \\
f(u_i) &= \begin{cases} 1, & i \bmod 2 = 0, \\ \frac{n}{4}+1, & i \bmod 2 = 1. \end{cases}
\end{aligned}
$$

Since

$$
wt(v_i) = \begin{cases} (\lfloor \frac{i}{2} \rfloor + 1) + 1 + 1 + 1 + 1 = \lfloor \frac{i}{2} \rfloor + 4 = \frac{i}{2} + 4, & i \bmod 2 = 0, \\ (\lfloor \frac{i}{2} \rfloor + 1) + (\frac{n}{4}+1) + (\frac{n}{4}+1) + (\frac{n}{4}+1) + (\frac{n}{4}+1) = \lfloor \frac{i}{2} \rfloor + 4 + \frac{3n}{4}, & i \bmod 2 = 1, \end{cases}
$$

$$
wt(u_i) = \begin{cases} (\lfloor \frac{i}{2} \rfloor + 1) + (\frac{n}{4}+1) + (\frac{n}{4}+1) + 1 = \lfloor \frac{i}{2} \rfloor + 4 + \frac{n}{2} = \frac{i}{2} + 4 + \frac{n}{2}, & i \bmod 2 = 0, \\ (\lfloor \frac{i}{2} \rfloor + 1) + 1 + 1 + (\frac{n}{4}+1) = \lfloor \frac{i}{2} \rfloor + 4 + \frac{n}{4}, & i \bmod 2 = 1. \end{cases}
$$

the weights of vertices of $W(3,n)$ under the labeling *f* constitute the set {4, 5, . . . , n+3} and the function *f* is a map from

$V(W(3,n)) \cup E(W(3,n))$ *into* $\{1, 2, \cdots, \frac{n}{4} +$

for *n* mod *4 = 0*.

Case 2.

$$
\begin{aligned}
f(v_iu_i) &= \begin{cases} \frac{i}{2}+1, & i \bmod 2 = 0, \\ \lfloor \frac{i}{2} \rfloor + 3, & i \bmod 2 = 1, \end{cases} \\
f(v_iu_{i+1}) &= \begin{cases} 1, & i \bmod 2 = 0, \\ 1, & i = 1, \\ \lceil \frac{n}{4} \rceil - 1, & i = 3, \\ \lceil \frac{n}{4} \rceil, & 5 \leq i \leq \frac{n}{2} - 2 \text{ and } i \bmod 2 = 1, \end{cases} \\
f(v_iu_{i+3}) &= \begin{cases} 1, & i \bmod 2 = 0, \\ \lceil \frac{n}{4} \rceil + 1, & i \bmod 2 = 1, \end{cases} \\
f(v_i) &= 1, \\
f(u_i) &= \lceil \tfrac{n}{4} \rceil + 1.
\end{aligned}
$$

Since

$$wt(v_i)=\begin{cases}(\frac{i}{2}+1)+1+1+1=\frac{i}{2}+4, & i \bmod 2=0,\\ (\lfloor\frac{i}{2}\rfloor+3)+1+(\lceil\frac{n}{4}\rceil+1)+1=\lfloor\frac{i}{2}\rfloor+6+\lceil\frac{n}{4}\rceil=6+\lceil\frac{n}{4}\rceil, & i=1,\\ (\lfloor\frac{i}{2}\rfloor+3)+(\lceil\frac{n}{4}\rceil-1)+(\lceil\frac{n}{4}\rceil+1)+1 & \\ =\lfloor\frac{i}{2}\rfloor+4+2\lceil\frac{n}{4}\rceil=5+2\lceil\frac{n}{4}\rceil, & i=3,\\ (\lfloor\frac{i}{2}\rfloor+3)+\lceil\frac{n}{4}\rceil+(\lceil\frac{n}{4}\rceil+1)+1 & \\ =\lfloor\frac{i}{2}\rfloor+5+2\lceil\frac{n}{4}\rceil, & 5\le i\le\frac{n}{2}-2 \text{ and } i \text{ m}\end{cases}$$

$$wt(u_i)=\begin{cases}(\frac{i}{2}+1)+1+1+(\lceil\frac{n}{4}\rceil+1)=\frac{i}{2}+4+\lceil\frac{n}{4}\rceil=4+\lceil\frac{n}{4}\rceil, & i=0,\\ (\frac{i}{2}+1)+1+1+(\lceil\frac{n}{4}\rceil+1)=\frac{i}{2}+4+\lceil\frac{n}{4}\rceil=5+\lceil\frac{n}{4}\rceil, & i=2,\\ (\frac{i}{2}+1)+(\lceil\frac{n}{4}\rceil-1)+(\lceil\frac{n}{4}\rceil+1)+(\lceil\frac{n}{4}\rceil+1) & \\ =\frac{i}{2}+2+3\lceil\frac{n}{4}\rceil=4+3\lceil\frac{n}{4}\rceil, & i=4,\\ (\frac{i}{2}+1)+\lceil\frac{n}{4}\rceil+(\lceil\frac{n}{4}\rceil+1)+(\lceil\frac{n}{4}\rceil+1) & \\ =\frac{i}{2}+3+3\lceil\frac{n}{4}\rceil, & 6\le i\le\frac{n}{2}-1 \text{ and } i \text{ mo}\\ (\lfloor\frac{i}{2}\rfloor+3)+1+(\lceil\frac{n}{4}\rceil+1)+(\lceil\frac{n}{4}\rceil+1) & \\ =\lfloor\frac{i}{2}\rfloor+6+2\lceil\frac{n}{4}\rceil=6+2\lceil\frac{n}{4}\rceil, & i=1,\\ (\lfloor\frac{i}{2}\rfloor+3)+1+1+(\lceil\frac{n}{4}\rceil+1)=\lfloor\frac{i}{2}\rfloor+6+\lceil\frac{n}{4}\rceil, & 3\le i\le\frac{n}{2}-2 \text{ and } i \text{ mo}\end{cases}$$

the weights of vertices of $W3$,n under the labelling f constitute the set

$$\left\{4,5,\cdots,3\lceil\frac{n}{4}\rceil+4\right\}\cup\left\{3\lceil\frac{n}{4}\rceil+6,3\lceil\frac{n}{4}\rceil+7,\cdots,4\lceil\frac{n}{4}\rceil+2\right\}$$

and the function f is a map from

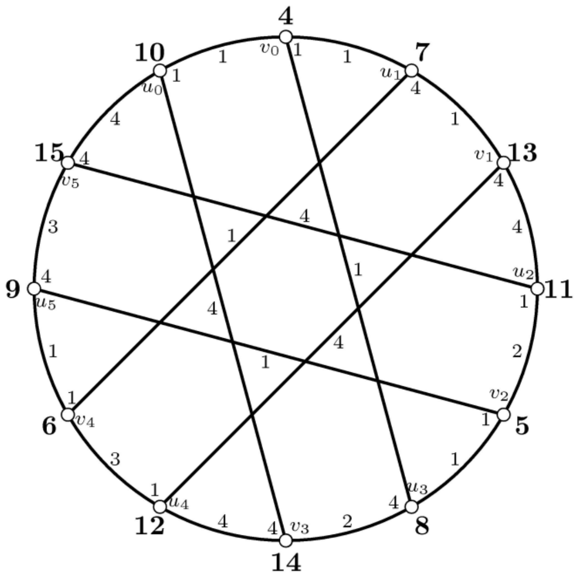
$V(W(3,n))\cup E(W(3,n))$ *into* $\{1,2,\cdots,\lceil\frac{n}{4}\rceil+1$

for $n \bmod 4=2$.

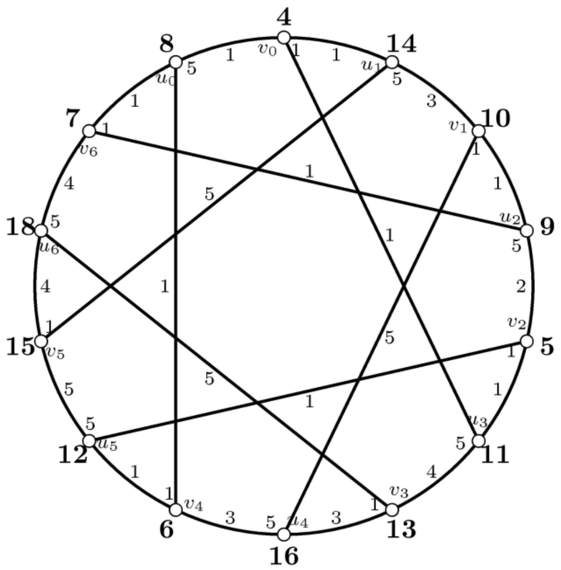
From case 1 and 2, the total labellings f has the required properties of vertex irregular total labelling. We then have $tvs(W(3,n.$ However, by Theorem 1.2,

$$tvs(G)\ge\lceil\frac{|V(G)|+\delta}{\Delta+1}\rceil=\lceil\frac{4\lceil\frac{n}{4}\rceil+3}{4}\rceil=\lceil\frac{n}{4}.$$ This concludes the proof. □

In Figure 2.2.2, we show the vertex irregular total labellings for W (3, 12) and W (3, 14).



W (3, 12)



W (3, 14)

Figure 2. 2. 2. Vertex irregular total labellings for W (3, 12) and W (3, 14)

3. Conclusion

In this paper, we apply the Bača's determination the bounds and precise values for some families of graphs concerning of parameters the total edge irregularity strength *tes(G)* and total vertex irregularity strength *tvs(G)*. We hope that the research along this direction can be continued, and in fact some results in this paper have already constituted a platform for further discussion concerning other graphs.

Acknowledgment

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REFERENCES

[1] M. Bača, S. Jendrol, M. Miller, J. Ryan, "On irregular total labellings", *Discrete Mathematics*, 307(2007), 1378–1388.

[2] J. L. Baril, H. Kheddouci and O. Togni, "The irregularity strength of circulant graphs", *Discrete Mathematics*, 304(2005), 1–10.

[3] S. Brandt, J. Mišskuf, D. Rautenbach, "On a conjecture about edge irregular total labeling", *Journal of Graph Theory*, in press.

[4] T. Bohman and D. Kravitz, "On the irregularity strength of trees", *Journal of Graph Theory*, 45(4)(2004), 241–254.

[5] J. Ivančo and S. Jendrol, "The total edge irregularity strength of trees", *Discussiones Mathematicae. Graph Theory*, 26(3)(2006), 449–456.

[6] S. Jendrol, M. TkáčThe "Irregularity strength of tK_p", *Discrete Mathematics*, 145(1995), 301–305.

[7] S. Jendrol, J. Miškuf and R. Soták, "Total edge irregularity strength of complete graphs and complete bipartite graphs", *Electronic Notes in Discrete Mathematics*, 28(2007), 281–285.

[8] S. Jendrol and V. ˇZoldák, "The irregularity strength of generalized Petersen graphs", *Mathematica Slovaca*, 45(2)(1995), 107–113.

[9] A. Meissner, M. A. Niwińska and K. T. Zwierzynski, "Computing an irregularity strength of selected graphs", *Electronic Notes in Discrete Mathematics*, 24(2006), 137–144.

[10] Khandoker Mohammed Mominul Haque, "Irregular Total Labellings of Generalized Petersen graphs", *Theory of Computing Systems*, 50(2012), 537–544.

[11] Kh. Md. Mominul Haque, "Irregular Total Labellings of Möbius Ladder", *SUST Studies*, to appear.

[12] O. Togni, "Irregularity strength of the toroidal grid", *Graphs and combinatorics*, 165/l66 (1997), 609–620.

[13] O. Togni, "Irregularity strength and compound graphs", *Discrete Mathematics*, 218(2000), 235–243.

Architecture of a Software Based Voting Management System for Bangladesh

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***Abstract*—Bangladesh has long suffered with its poll system. Erroneous vote counting, forged ballots, stolen ballot boxes are just a few of the many issues that are faced by the paper based ballot system. The Electoral System, a hardware based electronic voting machine, was invented by some brilliant people at the Bangladesh University of Engineering and Technology to combat some of these issues. Unfortunately, this system, with its limitations, could not counter all the issues with the paper based system. Closer inspection of the current system and studies carried out on existing systems in the developed democracies have led us to believe that a software based solution will be a better option for Bangladesh to gain the transparency and efficiency we need to achieve in carrying out elections. We propose such a system in this paper, which we believe will counter the numerous issues we face today with the election process. We have developed and tested part of this proposed system using Bengali at the user interface level.**

Keywords-poll; electronic voting machine; democracy; encryption, privacy

I. INTRODUCTION

Election is the key attribute of a democracy. People exercise their rights to choose who will govern the country through this popular system. But without proper handling of elections, a country will remain a democracy only in papers but not in practice. Several criteria have been mentioned in different literatures [5], which a voting system must employ to be effective. A voting system must be -

- Democratic - anyone eligible to vote must be allowed to vote.
- Private - it must not divulge, intentionally/accidentally, who a citizen has voted for.
- Accurate - a vote must be unalterable, non-eliminable

Elections have been less than effective in Bangladesh. Up until 2010, Bangladesh has been using paper based voting system. The system has some serious drawbacks [2][3][4], which include forgery, casting of multiple votes by the same person, ballot box looting, erroneous tallying and so on. In order to tackle these issues, a hardware based solution called Electronic Voting Machine (EVM) was introduced (Fig.1) by Pi Labs Bangladesh, Ltd.

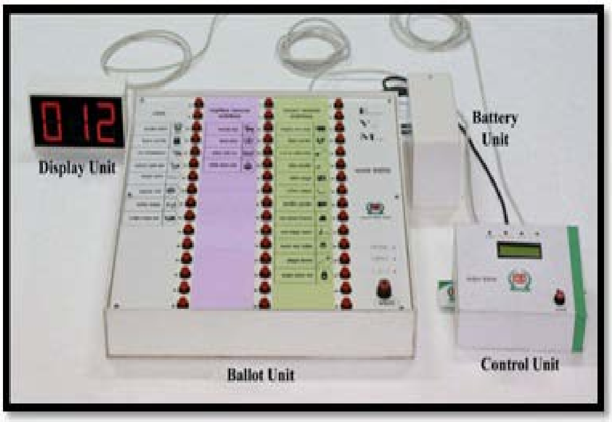

Fig. 1 EVM solution from Pi Labs, Bangladesh

This solution was first used in public in the 2010 Chittagong City Corporation election. It showed success in handling many of the issues encountered in the paper based ballot system. But unfortunately, the following issues still linger -

- No way of tracking if a person has already voted or not
- No way of checking identity of a person
- No way of creating a percentage count.
- Limitation on number of candidates that can be listed.
- Lacks flexibility in changing configurations.

In developed countries like the US, software based solutions[1] are used in polling stations. We have studied their implementations. We have come up with an architecture for an overall solution for election management system geared toward Bangladesh. We have also partially implemented the system as well using .NET/C# technology. This system, in its full implementation, will be able to replace the paper based and the hardware-alone based system that we have in use in Bangladesh.

II. OUR SOLUTION - ELECTION MANAGER

In order to counter issues faced by paper based ballot system and tackle problems unhandled by EVM, we have come up with a solution where software will play the key role in poll centers in Bangladesh. We have named our system Election Manager (EM). Components (Fig. 2) of the system are briefly discussed below -.

A. Central Server

A central system to manage all elections by appropriate authorities -

- Manage voter list
- Manage election
    - Manage candidate list
- Manage zones
- Tally final results with information collected from regional servers.

The central system will keep track of number of votes for each regional server. It will help in auditing to see if the total number of votes were manipulated or not.

B. Regional Servers

Regional servers will be needed at the zone level to collect vote from different poll center units. The servers will also monitor poll center units. It will keep a record of total vote cast for auditing purposes.

C. Clustered Database Servers

A central database where election results will be stored.

- A central database where election results will be stored
- It will be clustered for fault-tolerance and data safety.
- It will use open source database management system such as PostGreSQL.

D. Client Application for Poll Centers

The client application will reside in poll center units through which people will cast vote. This part has been implemented using .NET 4.5/C# technology with MSSQL as the local data store. We have used Bengali as the medium of the user interface.

III. SYSTEM OPERATION

Authorities will create election information using the central system using proper authentication. This information will be passed on to the regional servers

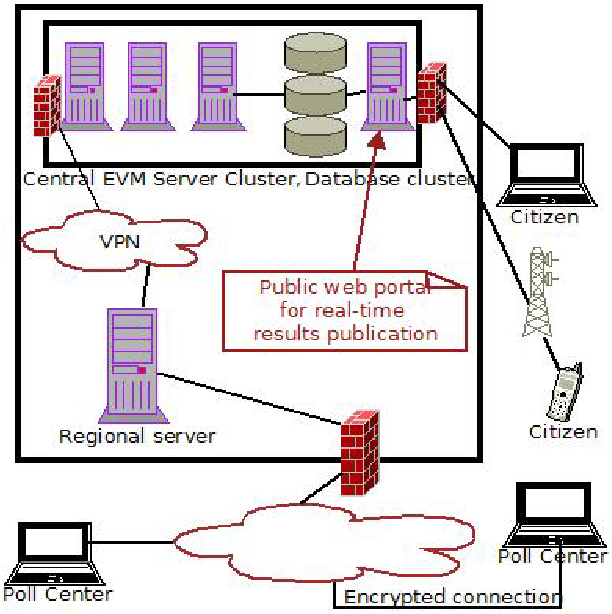

Fig. 2 EM system architecture

connected through Virtual Private Network(VPN). The regional servers will control the poll center client units, which can be laptops or customized kiosks. The individual client software will run on these client units. These units will be pre-configured to communicate with the regional server.

The regional servers will push down election information to the individual client units. The day before the election, in the presence of representatives from all parties, the unit will connect to the regional server and bring down the election information.

When a voter comes to poll center, the system will verify and validate a voter using a card reader to read from the voter id cards. The system will also check if the vote was already cast or not for this voter. Once authentication is done, the voter will be presented with the voting screen (Fig. 3) where she will be given choices to make. He/she will be assisted by audio.

Once the voter is done casting vote, two pieces of information, in XML format, will be sent to the regional server, which will then propagate the information to the central server. The information will include -

- who voted, so that the voter cannot cast vote anymore from anywhere.
- The vote information, which will include the poll unit's identification (listing 1), the zone information, and who the vote was for. It will not include the voter information to keep the vote private.

A copy of the vote will be kept in the local unit for verification purposes.

Identify applicable sponsor/s here. If no sponsors, delete this text box. (sponsors)

Listing 1. Message with vote information

```
<xml>
        <electioninfo infotype="vote">
                <election name="" type=""
date="" />
                <machineinfo id="" ip="" />
                <vote>
                        <!--candidate id will go
here-->
                        <votedfor
candidateid="" />
                </vote>
        </electioninfo >
</xml>
```

The central server will constantly monitor the health of the regional servers. The regional servers will monitor the health of the poll center units through ping or other utilities. If a unit fails to respond in a reasonable amount of time, the server will notify poll agents and administrators at the central station through SMS.

The central server will have the capability to remotely control regional servers. The regional servers will have the capability to remotely control the poll center units. This authority is needed to immediately shut down any unit where a problem may have occurred. It is also necessary to shut down voting process at the same time at the end of specified period to avoid any issues.

The central server will keep publishing results as they come in, in real-time to a web portal for public viewing.

IV. FUTURE WORK

The system is still under construction. We have listed the key areas that we have to work on-

- The server portion is partly developed where it can collect data from clients.
- The admin portal has to be completed
- Allowing people to vote from anywhere is yet to be added.
- The card reader is not integrated with the client side yet to automatically read voter information.

V. IMPLEMENTATION

The following is a not-so-comprehensive list of the resources we will need to implement and deploy the system -

- Hardware - multiple server machines for load balancing on the central server side, multiple database machines for clustering, firewalls, regional servers as needed

Fig. 3 Voting Screen

- Software - the client software that will allow people to vote, the central server that will accept votes from the poll centers, database software. We will also have to create the admin portal for managing elections by the election commission.
- Human resources - existing poll center agents can be utilized. We will need people to develop and maintain the system as well.

VI. SYSTEM SECURITY, SAFETY AND SCALABILITY

- Voter card reading will be added to avoid intentional/accidental identity issue.
- Fingerprint/face recognition will be added for securing the system with biometrics.
- Two level data security will be used -
  - Use two-level VPN to connect regional servers to the central server. This double encryption will provide extra security for data.
  - VPN is not feasible for connecting the vast number of poll centers units to the regional servers. This is why we have to use the following techniques to safeguard vote information being passed -
    - Regional server will accept connections from specific IP address or IP address range that will be assigned to poll center units.
    - Encryption must be used [5] -
      - The Mix-net Model
      - The Homomorphic Model

- The Blind Signature Model

VII. BENEFITS OF THE NEW SYSTEM

The following benefits of the system being developed can be easily surmised -

- Vote theft will become rare if not non-existent
- Identity theft will not be possible when biometric data will be used.
- Multiple votes cannot be cast by anyone.
- Will allow people to vote from anywhere to make sure everyone has an opportunity to vote.
- Keeping voting record at three levels (poll center units, the regional servers, and the central server) will allow auditing to find out any manipulation.
- Paper-less system
- Will create job opportunities for IT professionals and data entry people.
- Publishing of results in real-time.

VIII. CONCLUSION

In this paper we have introduced a software based solution to counter current issues we have with our polling system. We have implemented part of the solution. The other parts will be developed in phases. The solution, in its complete form, will save time, will create transparency in the polling process, will allow publishing results in a short amount of time. Duplicate voting will cease to exist, forgery will be burred. In the future, if we can integrate fingerprinting along with face recognition, this system will reduce any type of identity issue we face with current voting system.

REFERENCES

[1] R. M. Alvarez, J. N. Katz, C. Stewart III, R. L. Rivest, S. Ansolabehere, T. E. Hall, “Voting: What Has Changed, What Hasn't, & What Needs Improvement,” Caltech/MIT Voting Technology Project, August. 2011.

[2] Transparency International Bangladesh, “Transparency in Political Finance in Bangladesh,” The Crinis Pilot Project in the Asia Pacific, Oct. 2009.

[3] P. Eicher, Dr Z. Alam, J. Eckstein, “Elections in Bangladesh 2006-2009 - Transforming Failure into Success,” UNDP, 2010.

[4] S. M. Akram, S. K. Das, “Bangladesh Election Commission: A Diagnostic Study,” Transparency International Bangladesh, 2010.

[5] O. O. Olusola, O. E. Olusayo, O. S. Olatunde, G. R. Adesina, “A Review of the Underlying Concepts of Electronic Voting,” in Information and Knowledge Management, vol. II, 2012.

[6] T. Meisburger, “Strengthening Democracy in Bangladesh,” in Occasional Papers, The Asia Foundation, paper 13, June. 2012.

[7] [1] M. Boniface, “A Secure Internet-Based Voting System for Low ICT Resourced Countries,” M.S. thesis, IT, Makerere University, Kampala, UG, 2005

Exploiting GPU Parallelism to Optimize Real-World Problems

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***Abstract*— Construction of optimal schedule for airline crew-scheduling requires high computation time. The main objective to create this optimal schedule is to assign all the crews to available flights in a minimum amount of time. This is a highly constrained optimization problem. In this paper, we implement co-evolutionary genetic algorithm in order to solve this problem. Co-evolutionary genetic algorithms are inherently parallel in nature and they require high computation time. This high computation time can be reduced by exploiting the parallel architecture of graphics processing units (GPU). In this paper, compute unified device architecture (CUDA) provided for NVIDIA GPU is used. Experimental results demonstrate that computation time can significantly be reduced and the algorithm is capable to find some good solutions in a feasible time bound.**

Keywords: GPU;CUDA; Co-evolutionary genetic algorithm; Crew-scheduling; Min-max optimization.

I. INTRODUCTION

The scheduling of resources in the transportation industry is very complex and is a time consuming process. For instance, the airline industry faces largest scheduling problems in its daily operations [1]. The main problem every airline must solve is to construct an optimal schedule with the most efficient use of aircraft and crew resources in the timetable at a minimum cost to achieve maximum revenue. It is a real time optimization problem which should be solved in a given time to prevent any propagation of the disruption [2]. Therefore, these difficult optimization problems deserve a great deal of attention. However, these problems can be effectively solved by formulating min-max optimization problems.

Min-max optimization problems are found in different areas, for instance, in game theory, scheduling applications, network design, mechanical engineering, constrained optimization and function optimization [3]. These problems are considered difficult to solve deterministically in polynomial time bound. However, co-evolutionary genetic algorithms are found to be reliable to solve min-max optimization problems [4]. Co-evolutionary genetic algorithms usually operate on two or more populations of individuals. The populations evolve independently, but they are coupled together through fitness evaluation. The fitness of an individual in one population is evaluated on its performance against the individuals in the other population [5].

Using co-evolutionary genetic approach to solve min-max optimization problems requires high computation time to evaluate fitness of the individuals of each population [6]. Let us consider two populations each containing *n* individuals. The number of objective function calls required to evaluate each population is n^2. The individuals of one population must be evaluated against all the individuals with the other population. The evaluation of population needs to be performed in a number of times during the optimization process which is the most time consuming process [7]. Hence, this approach suffers from scalability problems if *n* is large or the objective function is complex. However, this scalability can be improved by exploiting the parallelism of GPU. In the proposed approach, fitness evaluation of each individual of one population against every individual in the other population is done in parallel. As a result, we can achieve a significant reduction in the computation time.

The remainder of the paper is organized as follows: in Section 2, the basic definition of min-max optimization problem is discussed. Section 3, describes the problem definition and co-evolutionary genetic algorithm. In Section 4, the implementation of the algorithm in CUDA architecture to solve the crew-scheduling problem has been discussed. Section 5 presents the experimental results and Section 6 concludes the paper.

II. LITERATURE REVIEW

Min-Max optimization problems allow one to find solutions by using scenarios to structure uncertainty [8]. In general, it can be defined as follows. Let's consider *X* is a set of all solutions and *S* is the set of all possible scenarios. If *F(x,s)* is considered to be the cost of a solution $x \in X$ in a scenario $s \in S$, then the task is to find some solutions which can minimize this cost over some scenarios. This is same as minimizing the maximum cost. According to this, the problem can be defined as follows:

$$\min_{x \in X} \max_{s \in S} F(x,s) \quad (1)$$

From [9], we find that the min-max problems were originally formulated by game theorists, which can be seen as an antagonist game where two players have a set of options. The player trying to find the solution *x*, tries to minimize the cost, while the player determining the scenario *s*, tries to maximize the cost. Later, these min-max problems were studied mathematically by many researchers. However, this problem is well suited with co-evolutionary genetic algorithms, since we can generate two different populations for *x* & *s* from two different search spaces.

III. Problem Definition & Algorithm

A. *Crew-Scheduling Problem Definition*

Given a set of trips that an agency must manage, the crew scheduling problem is to assign the trips to each crew in such a way that no trips are left unassigned in a given time. Let $\{T_1, T_2, T_3, \ldots\ldots T_n\}$ be the set of trips. Each trip T_i has a minimum processing time p_i and a maximum processing time q_i where $0 < p_i < q_i$. There are *m* crews $C_1, C_2, C_3, \ldots\ldots C_m$. We consider decision binary variables x_{ik} where,

$x_{ik} = 1$ [if T_i is assigned to crew C_k]

$x_{ik} = 0$ [if T_i is not assigned to crew C_k]

An assignment *x* is a feasible assignment (solution) if, for each trip T_i,

$$\sum_{k=1}^{m} x_{ik} = 1 \quad (2)$$

Let *X* be the set of all possible solutions. A scenario *s* is the combination of processing times. Thus $s = (p_1^s \ldots\ldots p_n^s)$. For each trip T_i, $p_i \le p_i^s \le q_i$, *S* is the set of all possible scenarios. *F(x,s)* is the time or cost of a solution *x* in scenario *s*. The cost is the maximum processing time to assign a trip to crew. Thus, we can define it as follows:

$$F(x,s) = \max_{1 \le k \le m} \left(\sum_{i=1}^{n} x_{ik}\, p_i^s\right) \quad (3)$$

Now the problem is to minimize this cost which can be formulated as min-max problems discussed earlier:

$$\min_{x \in X} \max_{s \in S} F(x,s) \quad (4)$$

We use co-evolutionary genetic algorithm to solve this problem.

B. *Co-evolutionary genetic algorithm*

The co-evolutionary genetic algorithm maintains two populations. The basic steps of the algorithm are illustrated below:

1. Initialize population *A* and population *B* at
   t = 0 / Initialize Populations */*
   / for i=1 to Maxitetrations */*
   / for j=1 to Maxgeneration1 */*
2. For each individual $x \in A(t)$, we evaluate
   $h(x) = \max(F(x,s): s \in B(t))$
   / fitness evaluation */*
3. Create new generation *A(t+1)* by reproduction, crossover and mutation
   / end of Maxgeneration1 for loop */*
   / for k=1 to Maxgeneration2 */*
4. For each individual $s \in B(t)$, we evaluate
   $g(s) = \min(F(x,s): x \in A(t))$
   / fitness evaluation */*
5. Create new generation *B(t+1)* by reproduction, crossover and mutation
   / end of Maxgeneration2 for loop */*
6. *t = t+1*, Unless *t* equals the maximum number of iterations go to step 2
   / end for loop of Maxiterations */*
7. Return the best solutions

From the Algorithm, we can observe that the number of evaluations of populations is *Maxiterations(Maxgeneration1 + Maxgeneration2)*. Considering *n* individuals in both populations, the number of evaluations of the objective function per populations is n^2. This evaluation is done in conventional CPU, which is also known as sequential evaluation. The formulation can be written as following:

$$Eval_{seq} = Maxit(Maxgen1 * n^2 + Maxgen2 * n^2) \quad (5)$$

For instance, if we consider *Maxiterations = 100, Maxgeneration1 = 10, Maxgeneration2 = 10, n = 50, we have* $Eval_{seq} = 5000000$. This computation is inevitable, but it is possible to evaluate the fitness of all *n* individuals in parallel. For this case, we can write the formulation as following:

$$Eval_{parl} = Maxit(Maxgen1 * n + Maxgen2 * n) \quad (6)$$

Considering the same parameters we have $Eval_{parl} = 100000$, which shows a clear advantage over sequential evaluation. Hence, we can obtain a significant reduction in computation by exploiting the parallel advantage of GPU.

IV. Implementation of the Algorithm using CUDA

A. *Basic Concepts of GPU & CUDA*

The Graphics Processing Unit (GPU) has emerged as a powerful computing device in this era of technology. The operational speed of GPU is much faster than the CPU (Central Unit Processing) [11]. The CPU mainly concentrates on arbitrary operations whereas on-the-other side GPU mainly concentrates on performance optimization related tasks. NVIDIA developed a software platform named compute unified device architecture (CUDA) for programmers to code in GPU. The language syntax consists of extensions of basic C-Language [10]. The depiction of basic CUDA architecture has been illustrated in Fig. 1. To support the parallelism of a program: threads, blocks and grids are used. In CUDA the functions are grouped into three categories: The *host* functions, which are called and executed only by the CPU. These functions are similar to those implemented in C. The *kernel* functions are only executed by the GPU device and called by the CPU. We have to use the qualifier, __global__, for this type of function. The return type of this type of functions is always void. Finally, there are device functions, which are both called and executed only by the GPU device. The qualifier, __device__, precedes the function definition for this type of functions. In case of device functions, it is allowed to return any type of value [11].

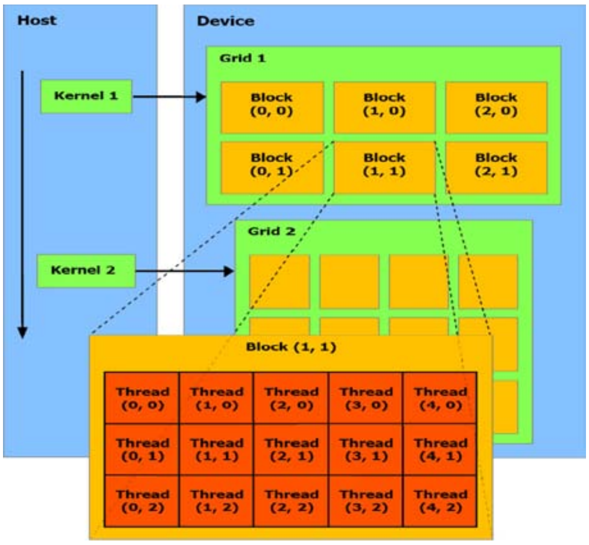

Figure 1. Basic CUDA Architecture

B. *Problem Formulation*

To evaluate the performance of this algorithm, first we try to find the worst-case scenario. For a solution, one of the worst-case scenarios is the one when all the trip assignments require maximum processing time. Thus, if we can reduce this worst-case scenario as much as we can, then we can create an optimal schedule. We define a lower bound on this algorithm. It is defined as follows:

$$L = \frac{1}{m}\sum_{i=1}^{n} q_i \qquad (7)$$

where q is the maximum processing time required for m crews. In addition, we formulate some problem sets to evaluate the performance of the algorithm. We consider two variables α_1 & α_2 to govern the processing time. The minimum processing time p_i is selected from a uniform distribution with the range [5, 20 α_1], where we set some arbitrary values for α_1. The maximum delay time d_i is selected from a uniform distribution with the range [0, α_2 p_i], where we set some arbitrary values for α_2. Hence, the maximum processing time is $q_i = p_i + d_i$. We try to find some good solutions closer to the lower bound as mentioned in Eq. (1).

C. *Implementation of the Algorithm in CUDA*

The main objective of this work is to minimize the processing time and generate an optimal schedule. In order to observe the robustness, we implement this algorithm in both sequential (CPU) and parallel (GPU) environment. The coding structure for these two environments is similar except the fitness evaluation and population generation part. Hence, in this section we discuss about these two parts regarding to GPU by which we can achieve parallelism. The algorithm first initializes the populations and then calculates the fitness in both randomly initialized populations. To note, we consider processing time as Population *A* and solution as Population *B*. Fig. 2 depicts the code which is responsible for invoking the kernel fitness function.

```
Fitness<<<nPop,nPop>>>(oldPopA,oldPop
B,nPop,nPar);
......................................
__global__ void Fitness(population A,
population B, int nPop, int nPar){
     __shared__ float fitness[NPOP];
    int popAid = blockIdx.x;
    int i;
    int popBid = threadIdx.x;
    float fit2 = mp( A[popAid].s,
B[popBid].g);
    fitness[popBid] = fit2;
        __syncthreads();
    if(popBid==0){
       float min = 100000;
       for(i=0; i<nPop; i++){
           if(fitness[i]<min){
              min = fitness[i];
           }
       }
       B[popAid].fit=min;
    }else if(popBid==1){
       float max = -100000;
       for(i=0; i<nPop; i++){
           if(fitness[i]>max){
              max = fitness[i];
           }
       }
       A[popAid].fit=max;
        }
}
```

Figure 2. Sample CUDA code for Kernel Fitness Function

The computation is done in parallel to calculate the fitness for each individual of populations A and B. Here each block indexed by blockIdx calculates the fitness of the individual in its corresponding threads indexed by threadIdx. This evaluation is done in two populations concurrently. However, there is some redundancy in calculations. This is because; each evaluation is done twice since shared memory is block-wise. We avoid this redundancy by utilizing global memory. Fig. 3 shows the corresponding CUDA code which is responsible for the generation of the new individuals of a population.

```
generateA<<<nPop,nPop>>>(newPopA,
oldPopA, oldPopB, cudaShuffle, nPop,
nPar, RAND, mut_rate, cross_rate,
bound, nRes);
………………………………………………..
__global__ void generateA(population
newPopA, population oldPopA,
population oldPopB, int* shuffled,
int nPop, int nPar, float* RAND,
float mut_rate, float cross_rate,
float *bound, int nRes){
    individual trialVector;
    int i, r1, r2, r3, j;
    __shared__ float fitness[NPOP];
    i = blockIdx.x;
    j = threadIdx.x;
    // uniform crossover
    if(j==0){

Crossover(oldPopA,newPopA,&trialVecto
r,i,nPar cross_rate, RAND);
……………
    }
    // mutation
    mutation(shuffled, nPop, &r1,
&r2, &r3, RAND);

makeTrial(oldPopA,&trialVector,r1,r2,
r3, mut_rate,nPar);
        __syncthreads();
    fitness[j]=proctime(newPopA[i].s,
oldPopB[j].g);
    __syncthreads();
    int k;
    float max = -100000;
    for(k=0; k<nPop; k++){
       if(fitness[k]>max){
           max = fitness[k];
       }
    }
    newPopA[i].fit = max;

    if (oldPopA[i].fit <
newPopA[i].fit){

       copyIndividual(&oldPopA[i],&ne
wPopA[i],nPar,nRes);
    }
}
```

Figure 3. Sample CUDA code for Kernel generation Function

The kernel function responsible for Population *A* generation and Population *B* is similar. Here, we represent for the case: Population *A*. The crossover needs an array to store the individuals to perform the operation and they are stored in the array named *trialvector*. This is generated by the host and indexed using the *threadId*. After mutation and picking the trial solutions the most fit values are stored in the required processing time by calling the device function *proctime()*. In this way, we can utilize the advantage of GPU for co-evolutionary genetic algorithm to ensure parallelism.

V. Experimental Results

A. Experimental Environment

To evaluate the algorithm performance we have carried out our simulations for sequential and parallel cases. To perform parallel evaluation we carried our experiment on linux server using NVIDIA GeForce GTX 580 driver which supports the CUDA compiler. The system specifications of the driver are listed in Table 1. To perform sequential evaluation we carried our experiment on Intel Pentium G620 processor working at 2.60 GHz clock speed. We adjust our machine to operate at 32-bit operating system including 8GB RAM memory.

Table 1. System Specifications for Parallel Evaluation

Parameter	*Values*
CUDA version	4.2
Global memory	3 GB
Local memory	48KB
Warp size	32
Maximum number of threads per block	1024
Maximum size of each Dimension of a block	1024x1024x64
GPU clock speed	1.54 GHz

Table 2, lists the parameters considered for co-evolutionary genetic algorithm.

Table 2. Parameters Considered for Co-evolutionary Genetic Algorithm

Parameter	*Values*
Maximum number of Iterations	100
Maximum number of allowable generations	100
Population size	50
Number of bits required by a string to represent an individual	64
Number of genes in a String	16
Number of bits restriced for a gene	4
Crossover rate	0.7
Mutation rate	0.003

Recall that the main objective of this work is to minimize the computation time by exploiting the GPU parallelism. Hence, we calculate the algorithm execution time in both CPU and GPU environments to observe the speedup. It is an indication of how much faster a parallel processing is over its counterpart, i.e., sequential processing [12]. The speed up is calculated by the following equation within the defined time bound.

$$Speedup = \frac{T_{seq}}{T_{parl}} \tag{8}$$

Here T_{seq} and T_{parl} are the total time taken by the sequential processing and parallel processing of the algorithm.

B. *Experimental Results*

To evaluate the algorithm, we compare our solutions to the lower bound as we defined earlier. In addition, the algorithm is able to create an optimal schedule by satisfying the requirements. In order to address this issue, we devise a schedule as illustrated in Table 3. In this schedule, we assume a normalized value to 1 to observe the performance of the algorithm. If the solutions can achieve closer to 1 then we can state that the algorithm have a good lower bound, and it can generate good solutions.

Table 3. Results for the Crew-Scheduling Problem

Values considered for governing the processing time	*Algorithm Performance for Sequential Evaluation (CPU)*	*Algorithm Performance for Parallel Evaluation (GPU)*
$\alpha_1 = 0.2$ $\alpha_2 = 0.6$	1.12	1.00
	1.15	1.01
	1.09	1.00
	1.08	1.03
	1.01	1.02
	1.07	1.03
	1.09	1.05
	1.02	1.00
	1.03	1.00
	1.10	1.02

From Table 3, we can observe that the algorithm performs well when it is performed in parallel situation rather than sequential evaluation. Here, we consider one problem set in which 10 instances are considered. In 1 instance, we consider a set of crews those are being assigned to their available trips in the processing time. The objective can be achieved by finding good lower bounds for the algorithm. Here, the values those are closer to the normalized value 1, are considered as good solutions. Hence, we can achieve an optimal schedule by minimizing the maximum processing time which exploits the parallel advantage of GPU.

The speed up achieved by the parallel evaluation is compared with sequential evaluation. To conduct this task, we fix every parameter except varying the generations. The results are depicted in Table 4. From Table 4, we can observe that, the computation time can be reduced significantly with parallel implementation to achieve robust solutions.

Table 4. Algorithm Execution Comparison

Number of generations	*T_{seq}[CPU (s)]*	*T_{parl}[GPU (s)]*	*Speedup*
20	6.721	0.099	67.88x
40	7.821	0.212	36.89x
60	10.38	0.582	17.83x
80	12.17	0.841	14.47x
100	14.69	1.061	13.84x

From Table 4, we can observe a reduction in speedup while we increase the generations. This is because; the search space increases while we increase the number of generations, which affects the algorithm execution time. The algorithm execution time is lower, when it creates an optimal schedule by finding good solutions in less number of generations compared to higher number of generations. Overall, the approach helps us to achieve our goal.

VI. CONCLUSION

This paper presents a co-evolutionary genetic algorithm to solve crew-scheduling problem by formulating min-max optimization problems. The main time consuming area of the co-evolutionary genetic algorithm is fitness evaluation of the individuals. In case of serial implementation, the evaluation of fitness function requires $O(n^2)$ time. The parallel evaluation of the fitness of the individuals brings down this time to $O(n)$. Finally, we can state that, the highly constrained real world optimization problems can be solved easily by inheriting the parallel advantage of GPU.

REFERENCES

[1] X. Chen, X. Chin, and X. Zhang, “Crew scheduling models in airline disruption management”, In Proceedings of the 17th IEEE International Conference on Industrial Engineering and Engineering Management (IE & EM), pp. 1032-1037, 2010. **(Conference)**

[2] T. H. Yunes, A.V. Moura, and C. C. d. Souza, “Solving very large crew scheduling problems to optimality”, In Proceedings of the ACM symposium on Applied computing, pp. 446-451, 2000. **(Conference)**

[3] T. Alamo, D. M. de la paena, and E. F. Camacho, “An Efficient Maximization Algorithm with implications in Min-

Max Predictive Control," IEEE Transactions on Automatic Control, vol. 53, no. 09, pp. 2192-2197, 2008. **(Journal)**

[4] K. Deb, S. Gupta, J. Dutta, and B. Ranjan, "Solving dual problems using a coevolutionary optimization algorithm," Journal of Global Optimization, 2012. **(Journal)**

[5] A. M. Cramer, S. D. Sudhoff, and E. L. Zivi, "Evolutionary Algorithms for Minimax Problems in Robust Design," IEEE Transactions on Evolutionary Computation, vol. 13, no. 02, pp. 444-453, 2009. **(Journal)**

[6] L. P. Veronese, and R. A. Krohling, "Differential evolution algorithm on the GPU with C-CUDA," In Proceedings of the IEEE International Conference on Evolutionary Computation, pp. 18-23, 2010. **(Conference)**

[7] A. K. Qin, V. L. Huang, and P. N. Suganthan, "Differential Evolution Algorithm With Strategy Adaptation for Global Numerical Optimization," IEEE Transactions on Evolutionary Computation, vol. 13, no. 02, pp. 398-417, 2009. **(Journal)**

[8] L. Liu, and Y. Zhang, "Gemetic Algorithm design of the Min-max weighted distance problem," In Proceedings of IEEE International Conference on Computer Application and System Modeling (ICCASM), pp. 620-623, 2010. **(Conference)**

[9] K. Masuda, K. Kurihara, and E. Aiyoshi, "A novel method for solving min-max problems by using a modified particle swarm optimization," In Proceedings of the IEEE International Conference on Systems, Man, and Cybernatics (SMC), pp. 2113-2120, 2011. **(Conference)**

[10] NVIDIA Homepage. Available: http://www.nvidia.com **(link)**

[11] C. C. Boyer, M. Meng, J. Tarjan, S. Sheafler, and K. Skadron, "A performance study of general-purpose applications on graphics processors using CUDA," Journal of Parallel and Distributed Computing, vol. 68, no. 10, pp. 1370-1380, 2008. **(Journal)**

[12] J. Verdu, A. Pauelo, and M. Valero, "The Problem of Evaluating CPU-GPU Systems with 3D Visualization Applications," IEEE Micro, vol. 32, no. 06, pp. 17-27, 2012. **(Journal)**

An Integrated Online Courseware Design Approach

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***Abstract*—E-learning helps people to acquire knowledge by using technology at any time and place. Courseware is an efficient way which covers most of the requirements of e-learning. Online courseware has become a demanding form of e-learning in present days. It helps to convert the conventional process of learning and teaching towards e-learning. This paper attempts to give an overview of some online renowned courseware. The authors make a comparison among the courseware and summarize them in the form of tree and table for easy demonstration of those. After studying the similarities and dissimilarities among various contents provided by those courseware, the authors propose a design approach for an integrated global online courseware which will combine all the facilities of the analyzed courseware into a single one. In order to accomplish the desired integration, authors have proposed a methodology with proper demonstration by Venn diagram. If the proposed global online courseware can be developed, then both the learners and instructors will be benefitted. Because of integrating all relevant contents to relevant courses, it will be more beneficial for the third world students and teachers also. Even though integrating all courseware contents depends on the permission of the concerned authority of the courseware, nevertheless global courseware pretends to be the necessity of modern era, especially for the self learners.**

Keywords— Courseware, E-learning, Global Courseware, Venn Diagram.

I. INTRODUCTION

With the rapid development of technology, e-learning pretends to be more useful and popular in education. E-learning means the use of electronic-media and ICT in learning and teaching. Electronic-media includes text, image, video, audio, animation etc. E-learning can be used in or out of the classroom. There are several forms of e-learning such as multimedia based f2f learning, online learning, distance learning, remote learning etc. Several institutions use e-learning in the form of courseware. Courseware is an additional educational software material that is used as kits for teachers or trainers and as tutorials for students, for use with a self-learning or coach assisted program.

There are several papers based on e-learning and courseware. L. Vcent *et al.* [1] conducted a survey upon the effects of multimedia lecture and traditional lecture. The result shows that multimedia processes for learning is better than traditional. But, bad functional technology disturbs multimedia process sometimes. F. Buendia *et al.* [2] proposed a system that allows students to send their system program using the web forms, execute them in a native operating system and receive their results and feedback information through web-browsers. L. Zhang *et al.* [3] propose an assessment model based on Bayesian networks, which assess learning status by knowledge map after absorbing and analyzing test results. A survey is conducted based on academic results and a questionnaire (MCQ) which shows that 1% student's attitude is negative that is, they are not satisfied. The authors have no way to know their mentality and what they actually require. A. Moini *et al.* [4] examines the problem of remote authentication in online learning environments and options of using biometrics technology to defend against user impersonation attacks by certifying the presence of the user in front of the computer, at all times. N. Hoic-Bozic *et al.* [5] describes a blended learning approach to course design and implementation using a LMS named AHyCo, applied to the senior students in the undergraduate program in a Mathematics and Information Science major at the department of Information Science. But, it is appropriate only where student/ instructor ratio is lower. J. R. Galvão *et al.* [6] analyzed various courseware features that are available on the internet and propose a new model that is based on some of the technologies, such as computers and Telecommunication.

The paper is structured as follows. In Section II, definition and types of courseware are described. In section III, present status of overall courseware is given with a brief overview of our analyzed courseware. Their tree and table representations are described in Section IV. In this section, a comparative analysis is made according to dissimilarity. In Section V, our proposed methodology for course design and development with application is explained. Benefits of the proposed idea and challenges to do this are given in Section VI and VII respectively. Finally, the conclusions and our contribution to this paper are summarized in Section VIII and IX correspondingly.

II. WHAT IS COURSEWARE?

All human beings have the right to learn, improve, and progress. Educational opportunity is the way by which the authors can fulfill that right. In this century, most of the persons related with education require delivering and acquiring knowledge with the use of technology. Courseware is an efficient way to do this. Courseware can include: (1) Material for instructor-led classes, (2) Material for self-directed computer-based training (CBT), (3) Web sites that offer interactive tutorials, (4) Material that is coordinated with distance learning, such as live classes conducted over the Internet, (5)Videos for use individually or as part of classes [7]. Technology is one of the biggest reasons for courseware. A courseware can be: (1) Visual – "hands on" type of person; Need pictures, graphs, etc. to learn and apply material, (2) Non-Visual - can apply learned material from just reading it. We can also characterize a courseware in the following sense: (1) On CD-ROMs – visual and customizable, (2) Books and Text – non-visual and very standardized, (3) On the internet – visual and accessible [8]. We are about to integrate the third type of courseware. There are different types of courseware user such as students, educator, self-learner etc. The MIT [9] open courseware conducts a survey based on who use their courseware and represents the result in a pie chart as fig. 1. The chart shows that the maximum number of courseware users is self learners. So, we are concerned

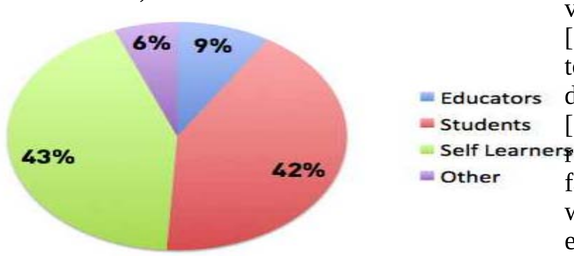

Figure 1. Pie chart of learners from MIT courseware

about designing an integrated courseware which will help all of the self learners. For a knowledge seeker, it will be time consuming to search all the individual courseware to acquire vast knowledge about a desired topic. But, if all the contents can be supplied by only a single courseware, then all the users will be benefitted.

III. PRESENT STATUS OF ONLINE COURSEWARE

We have analyzed several online courseware. Most of them are for undergraduate students. There are only a few for the post graduate students. Courseware for primary level students is even less. Khan academy [10] is one of this. No courseware is present for disable students. Different courseware have their own structure (not following any general structure). Courseware of different universities uses different books of different writers for providing contents to a course. There is an absence of content clustering technique for course material. For example, a course named 'Database System' may be taught from the book written by Korth or from the book written by Jeffrey Ullman or so on. Several books may contain several new topics and new methods for making a topic understandable on the same subject. In this case, existing courseware do not provide all topics from all books. But content clustering is very much necessary for self-learners. Though, there exist some attempts to integrate courseware, such as edx [11], open courseware consortium [12] etc. But, edx, combining different courseware, provides only some special courses. They also do not combine all courseware components, which may also be needed for a learner. They have combined courseware from several renowned universities such as MIT, Havard University, Berkely (University of California), The University of Texas System. So, here we do not get all courses in a single one. On the other hand, in open courseware consortium, they also provide some link of courseware for different courses. So, as usual we need to visit all courseware sites for acquiring knowledge.

We have analyzed 9 courseware provided by different reputed institutions. Different institutions present their courseware in different styles. All are designed to be appropriate in their own perspectives. MIT OpenCourseWare is a web-based publication of virtually all MIT courses (more than 2,000) content [9]. Open Yale Courses provide free and open access to a selection of introductory courses taught by distinguished teachers and scholars at Yale University [13]. Notre Dame OCW is a free and open educational resource provided by the Notre Dame University for faculty, students, and self-learners throughout the world [14]. Connexions is a place to view and share educational material made of small knowledge chunks called modules that can be organized as courses, books, reports, *etc.* [15]. Utah State OpenCourseWare is a collection of educational material used in the formal campus courses of Utah State University, and seeks to provide people around the world with an opportunity to access high quality learning opportunities [16]. Open Michigan, free learning from University of Michigan helps people find, use, and create openly licensed content and provides a space to share their educational content [17].

IV. A COMPARATIVE ANALYSIS

A. *Representation*

After analyzing all courseware, we make a tree representation of all of them to make them structured. A tree structure is a way of representing the hierarchical nature of a structure in a graphical form.

It is named a "tree structure" because the classic representation resembles a tree, even though the chart is generally upside down compared to an actual tree, with the "root" at the top and the "leaves" at the bottom [18]. A tree helps to represent hierarchical data. So, the tree representation established by us will help to easily understand a courseware. A tree shows the navigation through a courseware at a glance. At first, we make a tree representation of each courseware. When the tree is constructed, we can convert it to an equivalent table. The overview of courseware from Open Michigan University as a table got from tree representation is represented in fig. 2. Table representation is important, because it will help us to compare different courseware on the basis of some common criteria. We have found that about 50% of the courseware don't fulfill sufficient contents for a specific course. Some courseware provides unnecessary level or specification for getting a course material. Though so much specification may be time consuming and boring, nevertheless it helps to find a course material quickly. MIT is a vast courseware. But here, if the users can search course by department, then the 'find by topic' portion seems to be not required. In Notre Dame and Utah State University, there is no easy option of searching, like --search by dept. But, Notre Dame University provides bibliography for each course which can be useful for a knowledge seeker. Open Yale University provides support for buying books.

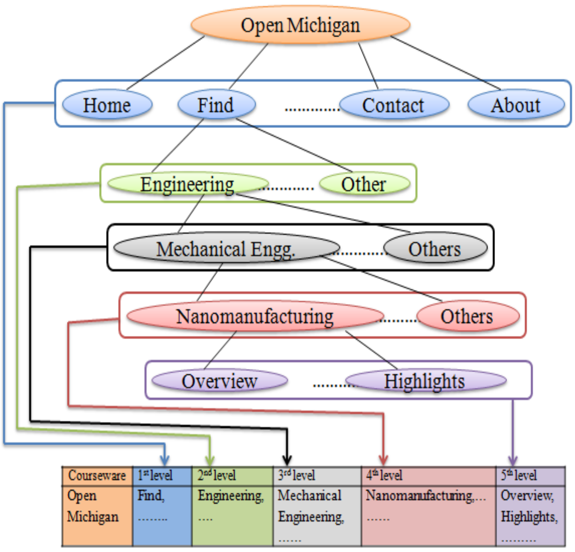

Figure 2. Overview of Courseware from Open Michigan University

Connexions has two important search options –search by Author & search by Keywords. Some institutions provide online courses only for some specific field. As for example, JHSPH provides courses related only health [19]. University of Washington [20] and CS50 [21] provide courses related only computer science. Now, if one equivalent table can be obtained from one tree, then the Table 1 can be found from observing other courseware. In this table, similar components are kept in the same level, even though they are not the member of that level. For example, courseware from Open Michigan has *Find* in the first level, but it has been kept in the 2nd level in table 1. In this table '…….' represents that there are more contents, which is not shown due to space congestion.

B. *Percentage of Similarity*

We have made a mathematical analysis for extracting the similarities among all courseware and express them in a percentage form. Every component of every level of each courseware is compared for similarity with every components of every level of all other courseware. In the present context, the number of comparison will be (9!), because here 9 different courseware have been analyzed. To make this comparison, we first assign a weight for each level of the courseware. Weights of levels are assigned according to their importance. Last level is the most important level, we want to consider. Since, it provides the desired courses and course materials. We primarily work for first seven levels. Assigned weights are shown in Table 1. For example, 1st level=5 in the first row, here 5 represents the weight of 1st level. Obtained percentage on the basis of similarity among courseware is given in the Table 2. For all diagonal elements, similarities will be 100%, because comparison occurs itself. If we consider for ID (1, 2) that 1st level of 1 contains 6 elements and 1st level of 2 contains 5 elements. There are 2 elements which are common in both ID 1 and ID 2. So, Similarity = (2*5)/9=1.11. Here, 2 is the number of common elements, 5 is the assigned weight and 9(6+5-2) is the number of elements which is found by applying union operation between the 1st level elements of ID 1 and ID 2. If all the elements of these two IDs in this level were matched, then their matching was assigned as 5. If a less matching is found, then a matching<5 is provided. For 2nd, 3rd and 4th levels, ID 1 contains value, but ID 2 not. So we can say that for these three levels, matching=0. Similarly, the values for other levels are measured. Now, for (1, 2) total similar weight = 1.11+0+0+0+0.1525+0.2376+3 = 4.5001. The values of other levels are also assumed. Now, percentage of weight = (4.5001*100)/56=8.04%. Here, 56 is the total weight (5+6+7+8+9+10+11=56). The percentage of this example is found manually. This technique has been applied in the program with real data and we find the table 2. After getting the results, we manually

Table 1. Tabular Representation of Different Courseware

ID	Courseware Origin	1st level=5	2nd level=6	3rd level=7	4th level=8	5th level=9	6th level=10	7th level=11	………. nth level=N
1	Massachusetts institute of technology	Courses …..	Find course by	Topic, MIT courseNo, **Dept**	School of science, -----	Biology------,(6)	Genetics -----	syllabus, lecture note, assignment, exam,------	
2	Utah state University	Courses ……				Anthropology,-------	Cultural anthropology	Syllabus, schedule, about professor,--	
3	Notre dame University	Courses ……	Areas of study			Aerospace and mechanical engg., ……	Thermo-dynamics, --------	lesson, bibliography, additional resources,---	
4	Open Yale university	Courses ……	View all courses	Dept, **course title**, course no, -----			Environmental politics and law, ………………	Session, survey, buy books, ………………	
5	Connexions	Content ……..	Search for content, **Browse content**	**Sub**, title, keyword, all collection ……….	**Arts,** Science ……….		Arts and culture, ………………...	E-mail, PDF, …………	
6	CS50		Search	Course name, ..			Mobile software engineering,----	Problem set, quizzes,-----	
7	Open Michigan	Connect ……..	Find		Engg, ……..	Chemical engg, ……	Process dynamics and controls , …….	Overview, highlights, …………	
8	JHSPH OCW	Course, topics, ………				Adolescent Health,… ……….	Child health & development, ……………	Syllabus, readings , ……………	
9	University of Washington	Courses					Computer programming ………………	Lecture , homework , ------------	

Table 2. Percentage of Similarities

ID	1	2	3	4	5	6	7	8	9
1	100.00%	4.99%	4.52%	4.44%	1.85%	0.56%	2.90%	2.96%	1.60%
2	4.99%	100.00%	8.71%	3.46%	1.98%	0.00%	3.35%	3.74%	0.89%
3	4.52%	8.71%	100.00%	5.80%	1.98%	0.00%	2.40%	2.75%	2.31%
4	4.44%	3.46%	5.80%	100.00%	0.89%	0.60%	3.77%	2.90%	1.16%
5	1.85%	1.98%	1.98%	0.89%	100.00%	0.00%	1.41%	0.00%	0.00%
6	0.56%	0.00%	0.00%	0.60%	0.00%	100.00%	0.56%	0.00%	0.00%
7	2.90%	3.35%	2.40%	3.77%	1.41%	0.56%	100.00%	1.64%	1.09%
8	2.96%	3.74%	2.75%	2.90%	0.00%	0.00%	1.64%	100.00%	1.16%
9	1.60%	0.89%	2.31%	1.16%	0.00%	0.00%	1.09%	1.16%	100.00%

check some of the percentages. We would never claim that these percentages are correct, because no synonym table is used. The mathematical calculation is correct, but when we are concerned with the terms like true positive, false positive, true negative and false negative, the result is too much incorrect. True positive is very weak, because without a synonym table, the program is unable to recognize that the course 'Basic Database' and 'Database System' are the same. So, the possibility of false negative is very high. Again, there may also be some words with same name but different meaning. For example, the spelling of a 'table' in database and a furniture 'table' are same, but their meanings are never the same. So, there may also some false positive. Even though we have used a program to compute the percentage, we had to do many works manually. The values in table 2 are represented by the column chart given in fig. 3. After analyzing the percentage of similarities, we have found that there exists less similarity among most of the courseware. Every courseware has at least a new idea. Some contents are important, but we see that one courseware have it, but others do not have. In

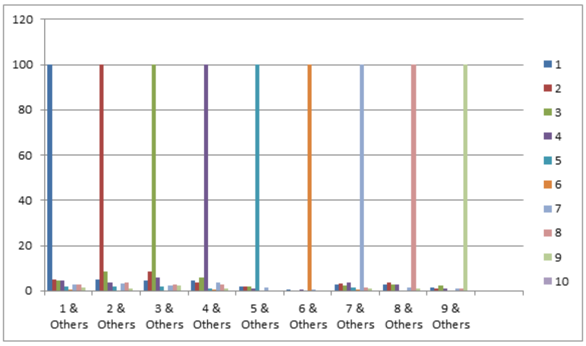

Figure 3. Percentage of similarity chart

this case, integration will combine all of the new ideas from different courseware.

V. Course Design & Development

The present era is an era of globalization. Globalization is the worldwide movement toward economic, financial, trade, and communications integration [22]. Now-a-days every conscious economic, financial, trade related institutes want to communicate with other similar organizations to get the advantages of globalization. There are different attempts to do this. BDREN [23] is one such attempt, which is an exclusive super highway communication network, linking education, research and innovation organizations in Bangladesh, and across the world. If we could combine the facilities of all of the courseware into a single one, then it can also be a step to globalization. We have already divided the course facilities into several levels through trees. Now if we can merge all the fields corresponding to a specific level of all courseware sites into a level of a single courseware so that it contains all the respective fields, then it can help different Universities to communicate with each other. It can also help a knowledge seeker to find the combined facilities into one.

A. Proposed Methodology

To design an integrated courseware, we have to compare each courseware with all other courseware. At first, we have to separate the similar and dissimilar contents of each level. Then we have to select which contents are to be kept in the integrated courseware. For this reason, firstly we will union the 1st and 2nd courseware from our analyzed courseware given in table 1, where 1 and 2 is the courseware ID. Here, we have considered similarity and dissimilarity of content of courseware ID 1 and 2. To represent dissimilarity we have used 1D and 2D and to represent similarity among 1 and 2 we have used S [1, 2]. To calculate union we will first add all contents of courseware ID 1 and 2. Then S [1, 2] is subtracted from the result of addition. The resulted output is represented by ‘a’ in fig. 4(a). Now, we will union the resulted output ‘a’ with courseware ID 3. Then we will repeat the same processes to find out the next resulted output. Now we represent the resulted output by ‘b’ in fig. 4(b). The resulted output ‘b’ will be union with the contents of courseware ID 4. In this step we will find ‘c’. By completing the described processes we will get the outputs ‘d’, ‘e’, ‘f’, ‘g’ and ‘h’ shown in fig. 4(d), 4(e), 4(f), 4(g) and 4(h) respectively. Here, ‘h’ is the desired unique value of analyzed 9 courseware. This operation will be conducted for each level separately. For each component found by union of one level, same operation will be used to find the next level for that component. For example, from the above calculation, we have found Home, Courses, About, Feedback etc. in the first level. Now, for each of these components same operation will be used to find the next important level. Among the dissimilar contents, there can have some synonyms. To make them computer understandable, we have to use a synonym table. When developing a synonym table, we face many problems. For example, in MIT, the course *Introduction to Psyschology* is taught in Fall 2011 and Fall 2004 with distinct topics. For the simplicity of synonym table, we compress these two courses into one. It has been done manually. An efficient synonym table is needed for an efficient integration of unique contents.

B. Application of Methodology

If we apply the methodology for each level stated in section V (A), the result will be all dissimilar contents with the similar contents only once. For example, we

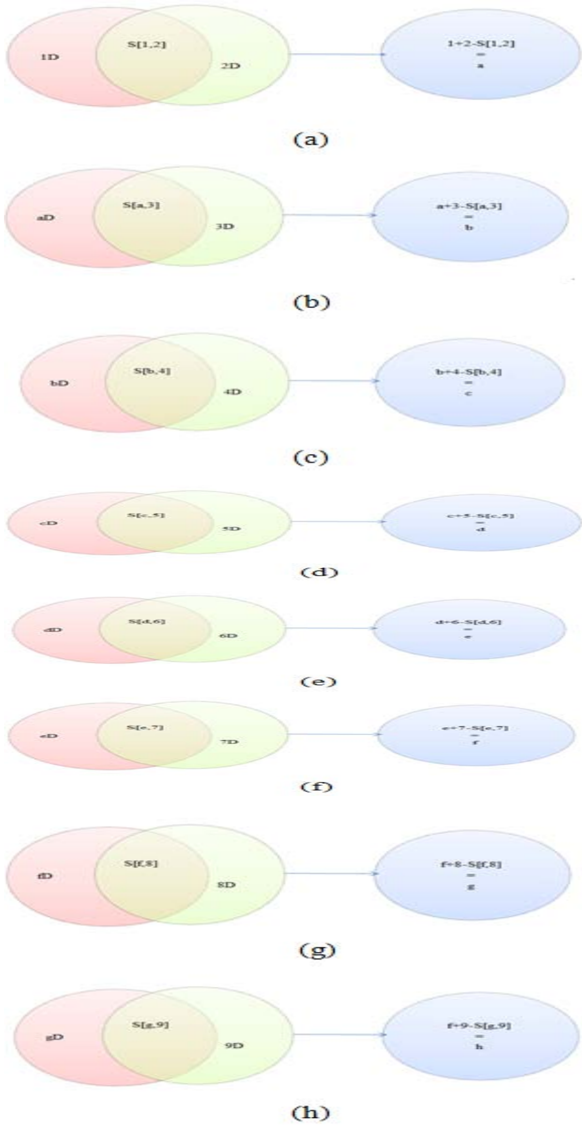

Figure 4. Representation of Methodology for Integrating Content

consider that only ‘Courses’ is found by applying the methodology in the 1st level. Now, MIT has Topic, Dept., and Course No as the next important level got from ‘Courses’ level and another courseware from Open Yale University provides Course No, Dept., and Course Title as the next important level of ‘Courses’ and so on for other courseware. Now, the result of applying methodology will be Topic, Course No., Dept., Course Title etc. as the next important level for the integrated courseware. Here, the important level means the level which we want to keep in our courseware as a level on the basis of their importance. For this reason, we have omitted the 2nd level of table 1. 2nd level is not so important for the integrated courseware to consider, because instead of using the contents of this level like ‘Find Course By’, we can go directly to the next level, which will be more easy and comfortable. Now, for all the components of the 2nd level, the same operation is done. Accordingly, after manual analysis of each level, we have made some summarization or component reduction such as Course No. It is University specific. So, in our proposed courseware, we do not want to keep this component. If a component seems to be not important from our perspective, then all elements under that component will be removed. From the collected data from different courseware and through the analysis of them in different manner, we have proposed an integrated global online courseware design approach for all kinds of learners. In fig. 5, we have made a tree representation according to our idea. Level wise components description for fig. 5 are given as: In the 1st level, we just want to keep the Union of the 1st level of all courseware like Home, Courses, About, Help, Feedback etc. From the components of 1st level, we have chosen Courses to represent our intention. In the 2nd level, we want to keep search option for courses according to school i.e. Science, Humanities, Business, Engineering etc. We have expanded the tree from Courses to the next level in the following figure on the basis of school. The 2nd level is expanded according to department of corresponding school. For the school of science, the department can be Chemistry, Physics, Mathematics etc. Other schools will also be expanded in the same manner. The 3rd level is expanded according to subjects under respective departments. For the department of Mathematics, the subjects can be Calculus, Differentiation, Digital Signal etc. The 4th level is expanded according to distinct sections under relevant subjects. For the subject Calculus, the distinct sections can be Single Variable Calculus, Calculus with Applications etc.

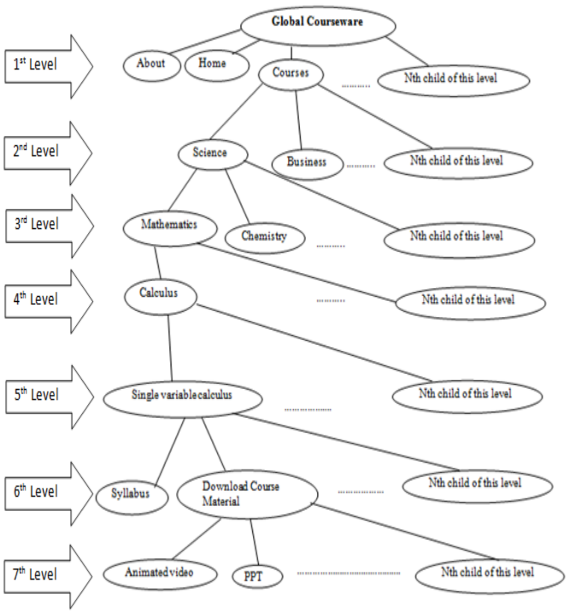

Figure 5. Tree Representation of Proposed Courseware

6th level will contain section wise components such as Syllabus, Download Course Material etc. required to get course related materials and information. In the 7th level, we will provide the entire course materials in the form of pdf, video, audio, animations etc as a child of Download Course Material. It is the most important level we are to consider. Learners will get all of their desired contents in integrated form from this level. We do not claim that only these levels are to be considered. Levels and components can be added or removed according to demand.

VI. BENEFITS

Different Universities have distinct courseware which are helpful for students of corresponding University. Different courseware have different good course material. In this case, searching for a content requires searching so much addresses to meet the desired need for a self learner. It is very much time consuming and difficult for him. But, if all of the accumulation of knowledge of various instructors from different universities can be found in a single one, then it will reduce the time and difficulties to search for contents. Also an instructor will be benefitted from this. He will find new material for self learning and teaching his students. For example, some courseware may provide only pdf for a course name ‘Algorithm’, another can provide the video file containing the lecture conducted by that University for that specific course, another can provide animation to make understandable different complex algorithms. If the integrated global online courseware can be developed, then all of these course materials will be found into a single page. So, we can claim that our proposed idea will provide greater benefits for modern age and it will surely be a step to globalization.

VII. CHALLENGES

To design an integrated courseware, we have faced some questions, such as: (1) Different University developed different contents for their courses. In many cases, the contents of a course vary sometimes on teacher, on level of universities, on prerequisite course. They have their own syllabus. Now question is ‘how to integrate this?’ (2) Another question arises for ‘who will manage these contents globally?’ (3) Third question is ‘how to get permission from concerned authority?’ We think that combining all courses and their contents is so tough through programming and manually, but not impossible. Managing contents globally is difficult. We assume that, any International organization (educational or others) can handle this. If the answers of the above questions can be given with proper solutions, then our proposed global online courseware will be the future necessity.

VIII. CONCLUSION

The purpose of authors is to develop an integrated courseware which will contain all courses of all courseware. If it can be completed, it will play an important role in e-learning. Instead of visiting all courseware, one can see all courses of every courseware by visiting only one site and this saves one's time. Though getting permission for merging contents of courseware from all Universities will be a tough matter, but, today or tomorrow, through the gradual evolution of globalization, it will be the necessity of the world. In different courseware, same course can be represented by different name. For this reason, all the calculations are not fully correct. To solve this problem, a proper synonym table will be helpful. To create a synonym table, authors have to do much work manually. Even after using this table, there exists much work which has to be checked manually. The proposed design approach is a continuous process. In order to make relevant changes in the functionality, comprehensive evaluation is needed.

IX. OUR CONTRIBUTIONS AND FUTURE RESEARCH DIRECTION

After analyzing several courseware, we have represented their status in an understandable manner. This data may be helpful for any researchers to make further investigation. We have proposed a method for integration of courseware contents, which will be helpful for the implementation. We have proposed only an outline for designing an integrated global online courseware, there is no implementation work. In the future, if all the concerned authority put their permission on developing this, then our proposed idea will completely fulfill the desire of all self learners.

REFERENCES

[1] L. Vicent, X. Àvila, J. Anguera, D. Badia, and J. A. Montero, "*Do Multimedia Contents increase the effectiveness of learning*?," 36th ASEE/IEEE Frontiers in Education Conference, October 28 – 31, 2006, San Diego, CA, pp. 12-17. **(Conference)**

[2] F. Buendía and J. Cano "*WebgeneOS: A Generative and Web-Based Learning Architecture to Teach Operating Systems in Undergraduate Courses*," IEEE Transactions on Education, vol. 49, no. 4, pp. 464-473 November 2006.**(Journal)**

[3] L. Zhang, Y. Zhuang1, Z. Yuan, G. Zhan, "*Auto Diagnosing: An Intelligent Assessment System Based on Bayesian Networks*", 37th ASEE/IEEE Frontiers in EducationConference, October 10 – 13, 2007, Milwaukee, WI , pp. 7-10.**(Conference)**

[4] A. Moini and A. M. Madni, Fellow, IEEE, "*Leveraging Biometrics for User Authentication in Online Learning: A Systems Perspective,*" IEEE Systems Journal, vol. 3, no. 4, pp. 469-476, December 2009.**(Journal)**

[5] N. Hoic-Bozic, Member, IEEE, V.Mornar, Member, IEEE, and I.Boticki, Student Member, IEEE, "*A Blended Learning Approach to Course Design and Implementation*," IEEETransactions on Education , vol. 52 , no. 1, pp. 19-30, February 2009.**(Journal)**

[6] J. R. Galvão& A. M. Barreto, Polytechnic Institute of Leiria, Leiria, Portugal,"*What is courseware? A comparative Analysis,*"World Transactions on Engineering and Technology Education Vol.4, No.2, 2005.**(Journal)**

[7] (2013, Apr) http://whatis.techtarget.com/definition/courseware**(link)**

[8] (2013, Apr) www.freequality.org/documents/training/**Courseware**%5B1%5D.ppt**(link)**

[9] (2013,Feb) http://www.ocw.mit.edu**(link)**

[10] (2013, Apr) https://www.khanacademy.org/**(link)**

[11] (2013, Apr) http://www.edx.org/**(link)**

[12] (2013, Apr) http://www.ocwconsortium.org/**(link)**

[13] (2013,Feb) http://oyc.yale.edu/**(link)**

[14] (2013,Feb) http://ocw.nd.edu/**(link)**

[15] (2013,Feb) http://cnx.org/**(link)**

[16] (2013,Feb) http://ocw.usu.edu/**(link)**

[17] (2013,Feb) https://open.umich.edu/**(link)**

[18] (2013, Apr) http://en.wikipedia.org/wiki/Tree_structure **(link)**

[19] (2013,Feb) http://ocw.jhsph.edu/**(link)**

[20] (2013,Feb) http://www.cs.washington.edu/**(link)**

[21] (2013,Feb) http://cs50.tv**(link)**

[22] http://www.businessdictionary.com/definition/globalization.html**(link)**

[23] (2013, Apr)http://www.bdren.net.bd/**(link)**

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Content-Based Image Retrieval using Haar Wavelet Transform

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Abstract—**Content-Based Image Retrieval (CBIR) system, also known as Query by Image Content (QBIC), is an image search technique to retrieve relevant images based on their contents. Here contents refer to the color, texture and shape of the image. In this paper we propose a content-based image retrieval system, where we use Haar wavelet transform to extract the image feature. We use f-norm theory to reduce the dimension of the feature vector and finally we use Canberra distance to calculate the distance between query image and database images. Our experiment result reflects the importance of Haar wavelet transform in CBIR system.**

Keywords— CBIR, Haar wavelet transform, Canberra distance, F-norm theory

I. INTRODUCTION

Content-based image retrieval (CBIR) [1] has been an active research topic in the last few years. Comparing to the traditional systems, which represent image contents only by keyword annotations, the CBIR systems perform retrieval based on the similarity defined in terms of visual features with more objectiveness.

A typical CBIR system automatically extract visual attributes like color, shape, texture and spatial information of each image in the database based on its pixel values and stores them in to a dissimilar database within the system called feature database. The feature data for each of the visual attributes of each image is very much smaller in size compared to the image data. The feature database contains an abstraction of the images in the image database; each image is represented by compact illustration of its contents like color, texture, shape and spatial information in the form of a fixed length real-valued multi-component feature vectors or signature. The users generally prepare query image and present to the system. The system automatically extract the visual attributes of the query image in the same mode as it does for each database image and then identifies images in the database whose feature vectors match those of the query image, and sorts the best similar objects according to their similarity value. During operation the system processes less compact feature vectors rather than large size image data therefore giving CBIR its cheap, fast and efficient advantage over text-based retrieval. Most CBIR systems work in the same way. A feature vector is extracted from each image in the database and the set of all feature vectors is organized as a database index. When similar images are searched with a query image, a feature vector is extracted from the query image and it is matched against the feature vectors in the index. Difference between the various systems lies in the features that they extract and the algorithms that are used to extract that features. The block diagram of basic CBIR system is shown in Figure 1.

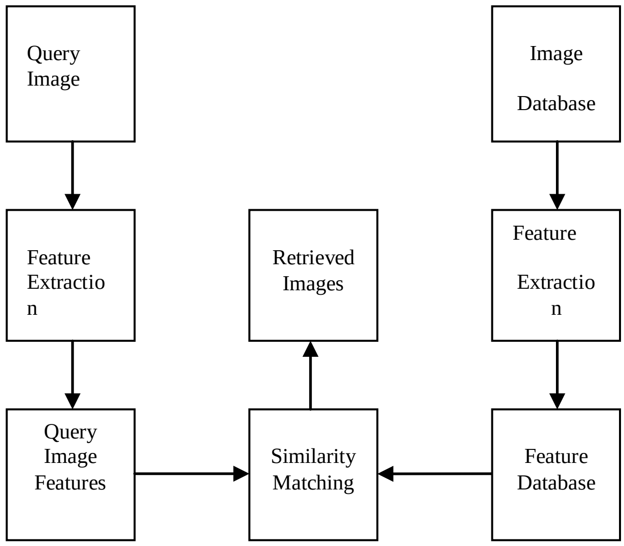

Figure 1. CBIR System

The significant applications for CBIR technology could be listed as art galleries [2], architecture design [3], medical imaging, criminal investigations, image search on the Internet [4, 5]. In this paper, we have proposed a content-based image retrieval system that extract image feature using Haar Wavelet transform, then we reduce the dimension of feature vector using f-norm theory, similarity between images is calculated by Canberra distance and efficiency of our proposed method is measured in terms of recall.

The rest of the paper is organized as follows. In section 2, we explain our proposed method. Section 3 describes the implementation and experimental results and finally conclusions are given in section 4.

II. PROPOSED METHOD

A. *Wavelet Transformation*

Wavelet transform provides a multi-resolution approach to texture analysis and classification [6]. The computation of the wavelet transforms of a two dimensional signal involves recursive filtering and sub-sampling. At each level, the signal is decomposed into four frequency sub-bands, LL, LH, HL, and HH, where L denotes low frequency and H denotes high frequency. Figure 2 shows the level 1 of 2D wavelet transform.

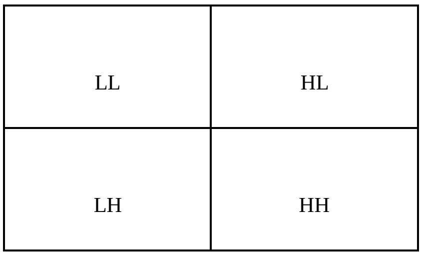

Figure 2. level 1 of 2D wavelet transform

We used Haar wavelet for image decomposition.

B. *Haar Wavelet Transform*

If a data set $X_0, X_1 \ldots. X_{N-1}$ contains N elements, there will be N/2 averages and N/2 wavelet coefficient values. The averages are stored in the first half of the N element array and the coefficients are stored in the second half of the N element array. The averages become the input for the next step in the wavelet calculation.

The Haar equations to calculate an average a_i and a wavelet coefficient c_i from an odd and even element in the data set are:

$$a_i = \frac{X_i + X_{i+1}}{2} \quad (1)$$

$$c_i = \frac{X_i - X_{i+1}}{2} \quad (2)$$

Steps for 1D Haar transform of an array of N elements:

1. Find the average of each pair of elements using equation 1. (N/2 averages)
2. Find the difference between each pair of elements and divide it by 2. (N/2 coefficients)
3. Fill the first half of the array with averages.
4. Fill the second half of the array with coefficients.
5. Repeat the process on average part of the array until a single average and a single coefficient are calculated.

Steps for 2D Haar transform:

1. Compute 1D Haar wavelet decomposition of each row of the original pixel values.
2. Compute 1D Haar wavelet decomposition of each column of the row-transformed pixels.

Red, green and blue values are extracted from the images. Then we apply 2D Haar transform to each color matrix.

Figure 3 shows the level 1 of 2D Haar decomposition of the image:

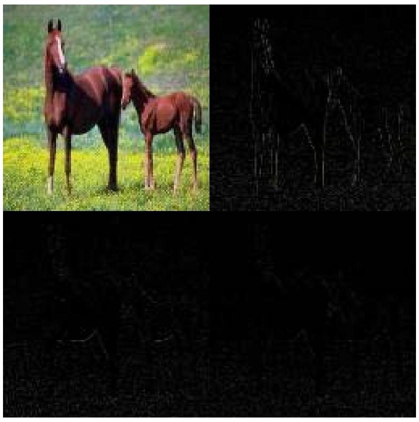

Figure 3. level 1 of 2D Haar wavelet transform

C. *Wavelet Feature Extraction*

We apply Haar wavelet decomposition of image in RGB color space. We continue decomposition up to level 4 and with F-norm theory [7] we decrease the dimension of image feature and perform highly efficient image matching.

Suppose A is a square matrix and A_i is its i^{th} order sub matrix where

$$A = \begin{bmatrix} a_{11} & \ldots & a_{1n} \\ \ldots & \ldots & \ldots \\ a_{n1} & \ldots & a_{nn} \end{bmatrix},$$

$$A_i = \begin{bmatrix} a_{11} & \ldots & a_{1i} \\ \ldots & \ldots & \ldots \\ a_{i1} & \ldots & a_{ii} \end{bmatrix} (i = 1 \sim n)$$

The F-norm of A_i is given as:

$$\left\| A_i \right\|_F = \left(\sum_{k=1}^{i} \sum_{l=1}^{i} \left| a_{kl} \right|^2 \right)^{1/2} \quad (3)$$

Let $\Delta A_i = \left\| A_i \right\|_F - \left\| A_{i-1} \right\|_F$ and $\left\| A_0 \right\|_F = 0$, we can define the feature vector of A as:

$$V_{AF} = \{\Delta A_1, \Delta A_2 \ldots \Delta A_n\} \quad (4)$$

D. *Similarity Measure*

The similarity between two images is computed by calculating the distance between feature representation of the query image and feature representation of the image in dataset. We use Canberra (eq.5) distance for distance calculation of the feature vectors.

$$dis(q,d) = \sum_{i=1}^{n} \frac{|q_i - d_i|}{|q_i| + |d_i|} \quad (5)$$

Where:

$q = (q_1, q_2 ... q_n)$, feature vector of query image

$d = (d_1, d_2 ... d_n)$, feature vector of image in database

n = number of element of feature vector.

If the distance between feature representation of the query image and feature representation of the database image is small then it is considered as similar.

III. EXPERIMENTS AND RESULTS

In our experiment, we select four types of image, 50 images in each category and 200 images in total from Wang's [8] database. The images are resized into 256x256.

The performance of a CBIR system can be measured in terms of its precision and recall. Precision measures the retrieval accuracy; it is ratio between the number of relevant images retrieved and the total number of images retrieved. Recall measures the ability of retrieving all relevant images in the database. It is ratio between the number of relevant images retrieved and the whole relevant images in the database. They are defined as follows:

$$\text{Precision} = \frac{\text{Number of relevant images retrieved}}{\text{Total number of images retrieved}} \quad (6)$$

$$\text{Recall} = \frac{\text{Number of relevant images retrieved}}{\text{Total number of relevant images}} \quad (7)$$

We extracted image features using Haar wavelet transform. Table 1 shows the average recall result for our proposed method.

Table 1. Average recall result

Category	Haar Wavelet Transform
Buses	0.71
Dinosaurs	0.68
Roses	0.79
Horses	0.67
Average Recall (%)	71.25

Figure 4 shows the graphical representation of our proposed method.

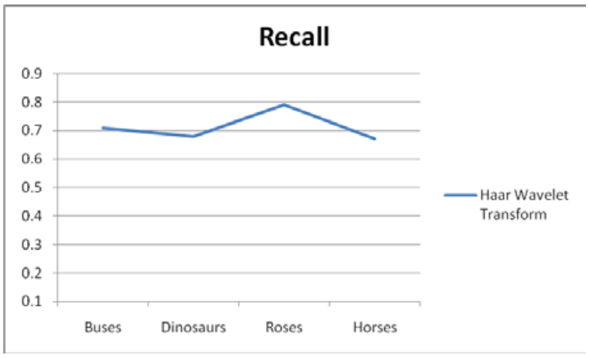

Figure 4. Average recall of proposed system

From Table 1 and Figure 3 it is seen that the average recall (%) based on our proposed method is 71.25. That means we can use Haar wavelet transform in CBIR system.

We implement our proposed method in Java. Some screenshot from our application for some sample query image is taken and they are shown below. Figure 5 shows retrieval result for the query image bus based on our proposed method. Figure 6, Figure 7, Figure 8 shows the retrieval result for the query images dinosaur, rose and horse respectively based on our proposed method.

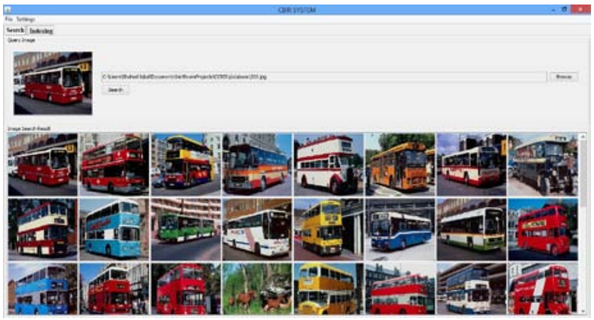

Figure 5. Retrieved images based on our proposed method for the bus as query image

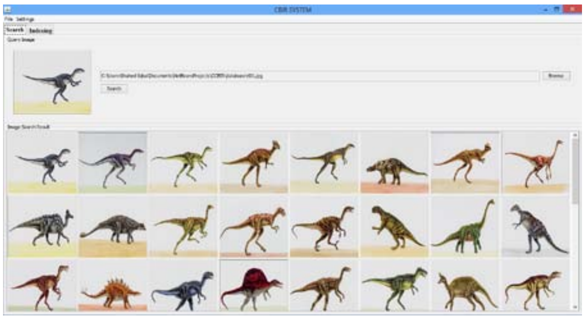

Figure 6. Retrieved images based on our proposed method for the dinosaur as query image

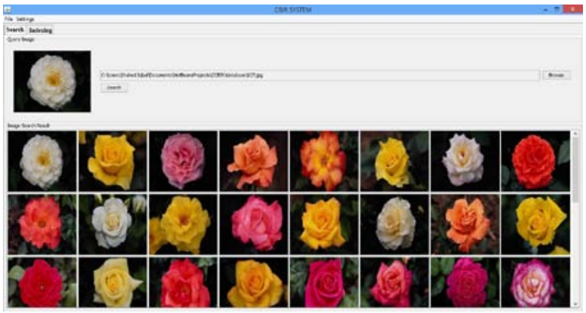

Figure 7. Retrieved images based on our proposed method for the rose as query image

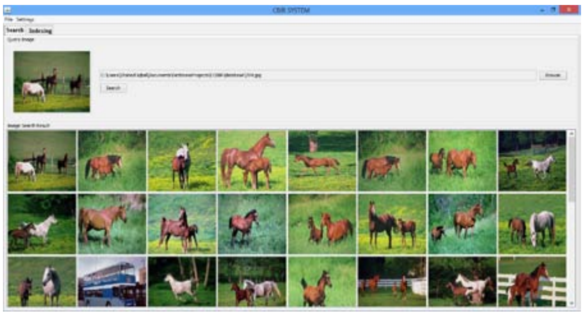

Figure 8. Retrieved images based on our proposed method for the horse as query image

IV. CONCLUSION

In this paper, we have proposed a CBIR method where we have successfully used the Haar wavelet transform. First we apply Haar wavelet transform to images in the database, then we apply F-norm theory to reduce the dimension of the feature vector and thus image feature is extracted using Haar wavelet transform. All the features for the database images are pre-calculated and stored in the database. Then when similar images are searched using a query image, we calculate the features for that image and the distance between the query image feature and database image features is computed using Canberra distance. Then we display the similar images according to their weight value. Our experiment result demonstrates that Haar wavelet transform can be used for image feature extraction in CBIR system.

REFERENCES

[1] R.Datta, D. Joshi, J.Li, J.Z. Wang, “Image retrieval: ideas, influences, and trends of the new age”, *ACM Computing Surveys*, vol. 40, no. 2, article 5, pp. 1-60, April, 2008.

[2] Dr. H.B.Kekre, Sudeep D. Thepade, “Improving ‘Color to Gray and Back’ using Kekre’s LUV Color Space”, *IEEE International Advanced Computing Conference 2009* (IACC’09), March 6-7 , 2009.

[3] Dr. H.B.Kekre, Sudeep D. Thepade, “Creating the Color Panoramic View using Medley of Grayscale and Color Partial Images ”, *WASET International Journal of Electrical, Computer and System Engineering (IJECSE)*, Vol. 2, No. 3, 2008.

[4] B.Ramamurthy and K.R.Chandran, “Content based Image Retrieval for Medical Images using Canny Edge Detection Algorithm”, *International Journal of Computer Applications*, vol.17, no.6, pp. 32-37, March, 2011.

[5] A. Ross, A. Jain and J. Reisman, “A hybrid fingerprint matcher”, in *proc. Of Int. Conf. on Pattern Recognition, Quebec City*, vol. 3, pp. 795-798, 2002.

[6] T. Chang and C.C.J. Kuo, “Texture analysis and classification with tree-structured wavelet transform, ” *IEEE Trans. on Image Processing*, vol. 2, no. 4, pp. 429-441, October, 1993.

[7] H. Huang, W. Huang, Z. Liu, W. Chen and Q. Qian, “Content-based color image retrieval via lifting scheme”, *Autonomous Decentralized Systems, 2005. ISADS Proceedings 2005*, pp. 378 – 383, 4-8 April, 2005.

[8] http://wang.ist.psu.edu/docs/related.shtml

Walking Steps Counting and Distance Measurement on User Smart Phone

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***Abstract*—As smart phone proliferates in our society all over the world, day by day it is becoming more familiar tool that is highly adapted in our everyday life. Most importantly these smart phones are equipped with various sensors that can be utilized to build up a wide variety of applications. In this paper, we present a method for counting the number of steps of a smart phone user, while walking at any variable speed. Hence, the cellular device must be configured with the acceleration and orientation sensor. For this purpose, the steps are detected based on a gravity with respect to time and the calculation of gravity is done from raw data of those sensors. The user's walking motion is recognized by android sensor (acceleration and orientation) and generated raw data. The walking distance with average step length 2.5 feet is assigned from survey study.**

***Keywords: Gravity; Raw-data; Sensor; Smartphone; Step length*.**

I. INTRODUCTION

In thousands of years ago, walk was called "man's best medicine" by Hippocrates who is the father of medicine, and the World Health Organization explicitly pointed out that "the world's best sport is to walk." in 1992 [16]. With the development of society, people pay more attention to making life healthy. Walking is not only effective to reducing body fat and improving muscle strength, but also it is simple and convenient for everyone to try without special equipment's. Walking 10,000 steps in a day is now widely recommended by physical experts as a target number for improving our health. For maintain physical fitness every person need effective walking. Walking is one of the most common forms of physical activity. It is impractical and uncomfortable to wear many sensors on specific locations in human body. For this reason we propose to use a mobile smart phone to monitor human walking activity. The attractive features of android devices are multiple different types of built in hardware that are accessible to developers. In additional, such hardware supported android is more suitable for creating creative applications than other smart hones [2, 16]. Android mobile phone platform is open enough to meet the requirements. The user's walking motion was detected by android built in sensor. The existing android platform sensor usually uses the single accelerometer sensor to count walking steps. According to the human walk process, android mobile phone establishes step counting model through analyzing the regularity of acceleration change in the gravity direction. Using only the acceleration sensor is difficult to measure the reflection based acceleration change in the gravity direction. For this purpose we need to combine the acceleration sensor with orientation sensor, in which orientation sensor maps acceleration to the gravity direction to achieve the accurate gravity acceleration change. The acceleration sensor with orientation sensor application analyzes the signal and calculates the walking distance. Regular exercise is necessary for healthy life and walking is the best job. Muscles and bones require burdening so that they will not become weak; their 'maintenance' is indispensable since the muscles hold and move the bones [5]. It is not possible for anybody to count 10,000 steps manually. The built in sensors make it possible for smart phones to recognize the kind of activity their users perform, for instance walking, jogging, cycling, working etc.

- Walking is a low impact, effective way to lose weight.
- Walking helps to reduce blood pressure.
- Walking is good for reducing the risks of heart disease and stroke.
- Walking improves the cardiovascular and pulmonary (heart and lung) fitness.
- Walking increases muscle strength, helps to create stronger bones, improves body balance and increases endurance.

How many steps do I need [9]?

Table 1. Step per day

For long term health and reduced chronic disease risk	**10,000 steps a day**
For successful, sustained weight loss:	**12,000 - 15,000 steps a day**
To build aerobic fitness:	**Make 3,000 or more of your daily steps fast**

II. General Idea of Sensor

The sensor is defined by the national standard GB7665-87 as: "Sensor can feel the specified measured signal which is converted into usable signal according to certain rules. Sensor usually is comprised of sensitive components and conversion components". Sensor is a detection device, which can feel the measured information, and can transform the information to electrical signal or other required formal information [2, 8]. Today's smart phones are programmable and come with a growing set of cheap powerful embedded sensors, such as an accelerometer, digital compass, gyroscope, GPS, microphone, and camera, which are enabling the emergence of personal, group, and community scale sensing applications. We believe that sensor- equipped mobile phones will revolutionize many sectors of our economy, including business, healthcare, social networks, environmental monitoring, and transportation [3]. Most android-powered devices have built-in sensors that measure motion, orientation, and various environmental conditions. The android platform supports three broad categories of sensors [12]:

- **Motion sensors**
  These sensors measure acceleration forces and rotational forces along three axes. This category includes accelerometers, gravity sensors, gyroscopes, and rotational vector sensors.
- **Environmental sensors**
  These sensors measure various environmental parameters, such as ambient air temperature and pressure, illumination, and humidity. This category includes barometers, photometers, and thermometers.
- **Position sensors**
  These sensors measure the physical position of a device. This category includes orientation sensors and magnetometers.

But different android versions have different built-in sensors [8].

A. *Acceleration Sensor*

Modern mobile phones are often equipped with acceleration sensors. The main effect of acceleration sensor is to obtain the movement of the mobile phone and this sensor acquires three parameters. They are the acceleration components which are separately equal to the accelerations of the coordinate system X, Y, Z axes minus the components of gravity acceleration in the corresponding axis [16]. The relationship between the acceleration coordinate system and mobile phone screen coordinate system is shown in Figure 1.

Figure 1. Acceleration Sensor Coordinates

3-Axis Accelerometer sensor gives us the acceleration measurements in m/s2 along each of X; Y; Z axes [10]. Different from the mobile phone screen coordinate system, the sensor's takes the left bottom as the origin, and the X axis is left to the right along the screen, the Y axis is down to up along the screen, and the Z axis is always perpendicular to mobile phone screen from down to up. The changes relationship of sensor data is as follows:
Mobile phone screen upward horizontal placed: (x, y, z) = (0, -0, 9.8); Uplift the top of the phone: the value of Y is reduced and is a negative value; Uplift the right of the phone: the value of X is reduced, and is a negative value; Mobile phone screen upward horizontal placed: z = approximately 9.8 [13].

B. *Orientation Sensor*

The main effect of orientation sensor is to obtain the orientation changes of the mobile phone and acquires three parameters. They are the rotation angles of Azimuth axis, Pitch axis and Roll axis. The relationship between the orientation coordinate system and mobile phone screen coordinate system is shown in Figure 2.

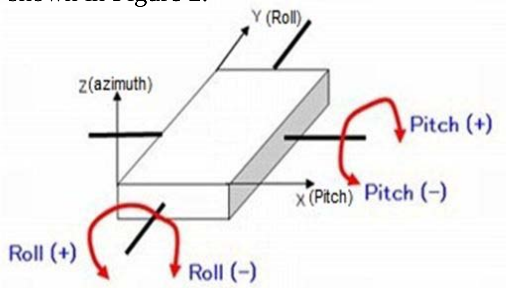

Figure 2. Orientation sensor coordinates

The direction of Yaw axis is the negative direction of gravity direction and it keeps unchanged. The direction of Pitch axis changes through the mobile phone rotating along direction of Roll axis is down to up along the mobile phone screen and unchanged [8].

III. RELATED WORK

Nowadays the sensor is more popular for monitoring human activity. Because many smart phones equipped with different types of sensors. Acceleration and Orientation sensor both measure more accurate value for counting step. Built in sensor in smart phone provides more facility for work with sensor. Though sometimes for noise sensor can't measure accurate raw data but it is more useful in our life for different purposes. Here we represent some related works such as–

(Pedometer and gait recognition) Pedometer is a popular device used to monitor the number of steps during the walking. They can be mainly divided into three types [7]. (Pedestrian Dead Reckoning System) The PDR system is used to estimate the route including the walking distance, heading direction, and even the change of height. It is mainly used to the location accuracy in an environment where GPS is not available [3]. Other related works are: author [7] discussed about "Daily walking distance questionnaire versus electronic pedometer". Researcher [4] worked on "Human activity recognition and energy expenditure generation system using motion sensors embedded in a smart phone (In a building environment)". In work, the "pedometer" which requires, practically, no user depended calibration and this scheme is based on an inductive proximity sensing principle [17]. Another interesting research on "A sensor-fusion-based wireless walking-in-place (WIP) interaction technique" is done for the same purpose of the present research [1].

After analyzing the previous works we have got many limitations such as they prefer more importance on pedometer than mobile application. Smart phone is more portable than pedometer today. Beside this, smart phone is a fundamental element in our daily life so that we have to bear it all time. Therefore, a built-in sensor on smart phone is investigated for step counting and distance measurement in this present research.

IV. METHODOLOGY

Step counters are becoming popular as an everyday exercise measurer and motivator. Accelerometers are becoming increasingly ubiquitous in commercially sold devices, such as mobile phones like the Apple i-Phone or the Samsung Galaxy Ace. But not only using the acceleration data the result of detection steps or distance is not so much accurate. If we combine acceleration sensor with orientation sensor the result will be more accurate [2]. Here we proposed step detection algorithm that utilizes the accelerometer of a waist-mounted smart phone. And also we calculate distance between starting state and ending state of walking. Modern mobile phones are often equipped with acceleration sensors. Considering the fact a mobile phone can be at any position on the user body, with time-varying orientation change, we use the magnitude of 3-axis accelerometer reading instead of its vertical and horizontal components [2]. In Orientation sensor also have three axes. The three axes are azimuth, pitch and roll. Azimuth is used to detect the compass direction, where 0 = North, 90 = East, etc... Pitch is used to track the devices side to side orientation, where vertical is around 0, tilted so the left side is on top is around 90, and tilted so the right side is on top is around -90. The documentation for pitch says the values range between -180 and 180 [3]. The roll represents the degree to which the device is tilted back and forth. A more promising possibility for measuring distances is counting steps based on the readings of the acceleration sensor and orientation sensor. By counting steps, distances cannot be measured exactly, but good estimates can be made. Multi-sensor pedometer application makes acceleration mapped to the gravity direction using orientation sensor, and gains accurate gravity acceleration changes [14].

The steps feature extraction of the single acceleration sensor is to analyze the absolute values of the acceleration to determine the wave crest (wave troughs). The acceleration values were recorded in meters per second squared (m/s2). The definition of acceleration the absolute values are as follows [15]:

$$G_a = \sqrt{(X^2+Y^2+Z^2)} \quad (1)$$

Here G_a is the gravity for acceleration sensor. We can count step only this gravity value. But the result will not so much accurate. Because user can't always keep still mobile phone, they can also rotate the device as necessary.

In Orientation sensor also have three axes. The three axes are azimuth, pitch and roll. The orientation values provide us the rotation angle of the device. So the gravity for orientation sensor is as follows:

$$G_o=\sqrt{((\sin(roll))^2+(\sin(pitch))^2+(\sin(azimuth))^2)} \quad (2)$$

But only the gravity for orientation sensor can't count the step because it show only rotation angle, for step counting we also need change of position.

After analyzing the sensor equations 1 and 2, the proposed method is applied for step counting. In this approach, initially calculate gravity a_x, a_y, a_z. Then calculate the total gravity G_v. G_v is the finding gravity and final result depends on this gravity and the threshold value. If we calculate real time threshold value then we must get more accurate value than to use one sensor value. Here, take components of X, Y, Z for acceleration and roll, pitch for orientation sensor, in gravity direction respectively as the variable a_y, a_x, a_z.

a_x= X ×Sin(roll) (3)

a_y= Y ×Sin (pitch) (4)

a_z= Z× Cos (roll) × Cos (pitch) (5)

G_V=√ (a_x+ a_y+ a_z) (6)

The formula for calculating the Real-time dynamic threshold- "Th" is as follows:

(7)

When the value of the crest G_V > Th, the crests is effective and the count of steps should be added 1.

A. *Step Counting Process and Distance Measurement*

1) Device positioning and orientation

The device is fixed near the center of mass of the subject, as shown, so that the net external force acting on it can be detected by the accelerometer of the device. The positive x, y, and z axes point to the left of the subject (horizontal), vertically upwards and backwards (horizontal) for the subject respectively. The y-direction acceleration data shows the strongest amplitude response, which could be more useful to detect human walking step.

2) Steps counting

The step counting process with a flowchart is given in figure 3. Now step by step process is given below:

- In the starting point, taken the value for the sample number of walking steps that created the waveform. The variable "i" is the current sample number of walking steps.
- To calculate threshold value "Th".
  There are two methods to determine threshold [11]. One way is by the number of tests gravity data with fixed threshold, and the other method is through the real-time data collection. For real-time data collection calculate "Th" with n number of sample and store it database.

- Got gravity data[i] and next gravity data[i+1].
- Compare data[i] and data[i+1].
- If data[i] and data[i+1] is not equal then compare data[i] with threshold value "Th".

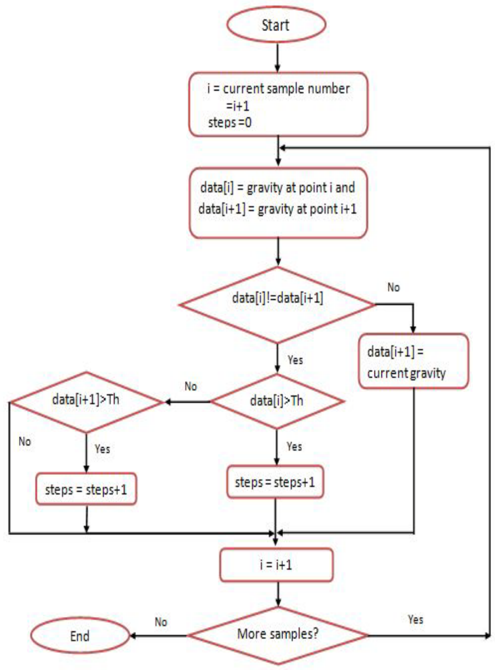

Figure 3. Flow chart of Step Counting Process

- If data[i] is greater than "Th" then one step count.
- Compare data[i+1] with "Th" if data[i] is less than "Th".
- If data[i+1] is greater than "Th" then one step count otherwise increase sample number and calculate with gravity for this sample number.
- When data[i] is equal to data[i+1] then we replace current gravity with data[i+1] and increase the value of sample number.

To continue this process until the total walking steps is considered.

3) Distance Measurement

The step length is varying from person to person. The research level information is that the average person's stride length is approximately "2.5 feet" long [6]. Yet we also have made a survey for finding average step length and represent this survey data with a line chart in Figure 4. With this data we can calculate the distance of human walking.

Distance = Total Step*(2.5feet or 0.732m) (8)

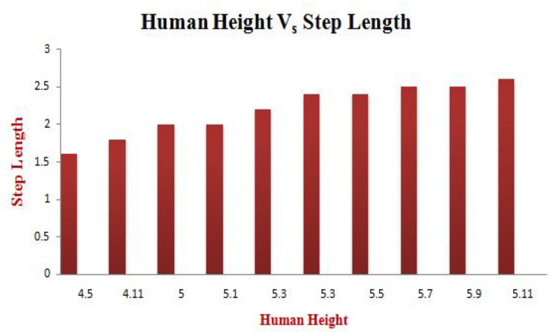

Figure 4. Survey chart

The walking distance with average step length 2.5 feet is assigned from survey study.

B. *Difference with existing system*

The more familiar step counting existing system is pedometer (is a device). Now-a-days smart phone is also equipped with pedometer. Pedometer is popular for step counting, measurement of walking distance and energy loss. But our approach work for smart phone user's and cellular device must be configured with acceleration and orientation sensor. These two sensors collect raw data and this raw data is for equation (6). When equation (6) executed we get G_v.

V. Implementation and Result Analysis

In implementation, acceleration sensor is considered to calculate G_v in android platform. Android is an open source operating system for smart phone. Implementation emulator is demonstrated the axes acceleration value in figure 5. It showed three axes acceleration value and calculated the value of G_v. A line chart illustrate for the result of change of three axes acceleration value.

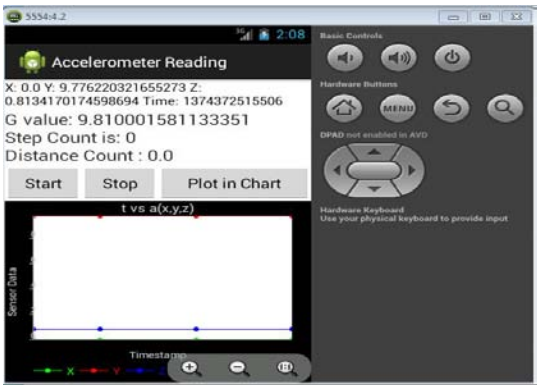

Figure 5. Emulator Output

In programming, a built in function namely sensor manager is used for taking sensor value.

```
sensorManager =
(SensorManager)getSystemService(Context.SENSOR_SERVICE);
```

The acceleration sensor is taken value by following code:

```
Sensor accel =
sensorManager.getDefaultSensor(Sensor.TYPE_ACCELEROMETER);
```

In this method, acceleration and orientation sensor is used to count steps and calculate the distance with average step length. The world wide accepted step length is "2.5 feet" that equal to our survey result. Raw data is considered of those sensors for measurement of gravity G_v. Now represent this gravity G_v with a sine wave in figure 6.

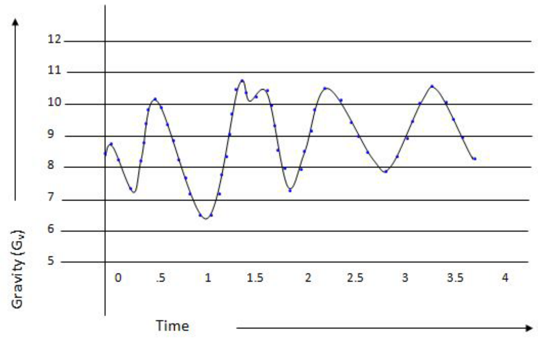

Figure 6. Waveform of Gravity (G_v) with respect to time

Hence, it is illustrated the different values of gravity with respect to time and calculated the real-time threshold value "Th" from real time threshold equation. At the time of 0.5 second the respective value of G_v is as follows:

8.4+ 8.7+ 8.5+ 7.3+ 8.2+ 8.8+ 9.5+ 9.8 +10.2 +9.9 = 89.1

Now, calculate threshold value for 10 samples.

Th = (89.1/10) + 1 = 9.91

To calculate the total number of steps compared this threshold value with corresponding gravity. If gravity is greater than "Th" then count one step. From Figure 6, we have got 1 step in 0.5 second because only one sample is greater than 9.91.

VI. Conclusion

User's walking motion is detected by acceleration sensor and orientation sensor. The combinations of those sensors are used to find more accurate result than the single sensor application. The advantage of multi-sensor is more accurate than the single sensor application. This research has been done the implementation and the results analysis of the human step counting with a smart phone. Hence, single acceleration sensor equation is considered to do the work for step counting. On the other hand, without jerking, this system does not give any output. Besides all smart phones does not contain all types of sensors. This work might be extended bearing in mind the real-time threshold value in the android application.

Acknowledgment

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REFERENCES

[1] Ji- Sun Kim*, Denis Gracanin, Francis Quek "Sensor- Fusion Walking-in –Place Interaction Technique using Mobile Devices", IEEE Virtual Reality 2012, 4-8 March, Orange County, CA, USA, 978-4673-1246-2/12/$31.00 ©2012 IEEE.

[2] W u, Shyi-Shious, Wu,Hsin-Yi "The Design of an Intelligent Pedometer using Android" , 978-0-7606-3/11 $26.00 ©2011 IEEE, DOI 10.1109/IBICA.2011.83.

[3] Nicholas D. Lane, Emiliano Miluzzo, Hong Lu, Daniel Peebles, Tanzeem Choudhury, and Andrew T. Campbell, "A Survey of Mobile Phone Sensing" IEEE Communications Magazine, September 2010.

[4] Jin-Ho Cho, Jeong Tai Kim, Tae- Seong Kim "Smart Phone- based Human Activity Classification and Energy Expenditure Generation in Building Environments", IEE EMBS Conference 2010; 5172-5175.

[5] Gobor Kiss "Using smart phones in healthcare and to save lives", 978-0-7695-4580-6/11 $26.00 ©2011 IEEE.

[6] Liang-Yu Chen, Wen-Yuah Shih, Kun-Chan Lan " Estimating walking distance with a smart phone" department of CSIE, National Cheng Knug University,2006

[7] David R. Bassett, Jr Andra L. Cureton, And Barbara E. Ainsworth "Measurement of daily walking distance-questionnaire versus pedometer" Exercise Science Unit, The University of Tennessee, Knoxville TN; and Department of Epidemiology and Biostatistics & Department of Exercise Science, University of South Carolina, Columbia SC, Analog dialogue 44-06, June 2010.

[8] Shahid Ayub, Alireza Bahraminisaab, Bahram Honary "A Sensor Fusion Method for Smart phone Orientation Estimation" ISBN: 978-1-902560-26-7 © 2012 PGNet.

[9] Melis oner, Jeffry A, Pulcifer-Stump, Patrick Seeling "Towards the Run and Walk Activity Classification through Step Detection—An Android Application", 98, 208101-4, 2004.

[10] Baek, J,-H., Park, W.-B., Yun, B.-J. "Accelerometer signal processing for user activity detection", Lecture Notes in Computer Science (Springer) 2004; 3215: 610-617

[11] Michal Tomlein, Pavol Bielik, Peter Kr´atky, Stefan Mitr´ık, Michal Barla and M´aria Bielikov´a "ADVANCED PEDOMETER FOR SMARTPHONE-BASED ACTIVITY TRACKING", VG1/0675/11, KEGA 028-025STU-4/2010.

[12] Khan, A. M., True, P.T. H., Lee, Y.-K., Kim, T.-S.: A tri- axialalerometer- based physical- activity recognition viaaugmented- signal features and a hierachical recognizer; IEEE transactions on information technology in biomedicine (IEEE EMBS)2010; 14:1166-1172

[13] Mark Pedley "Tilt Sensing Using a Three-Axis Accelerometer" Document Number: AN3461 Rev. 6, 03/2013.

[14] D.M. Karantonis, M.R. Narayanan, M. Mathie, N.H. Lovell, and B.G. Celler, "Implementation of a real-time human movement classifier using a triaxial accelerometer for ambulatory monitoring", IEEE Trans. Inf. Technol. Biomed., vol. 10,pp. 156-167, jan.2006.

[15] Sauvik Das, LaToya Green, Beatrice Perez, Michael Murphy, "Detecting User Activities using the Accelerometer on Android Smart phones" July 30, 2010.

[16] Song Haoran ,Liao Wenshuai, Zhao Yiming, " Based on the ADXL330 Acceleretion Sensor Precision Pedometer" , August 2006.

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Sentiment Mining in Social Network Using Textual Opinion

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Abstract—Opinions which are collected from social networks are the great resource for analyzing public sentiment for different purposes. Sentiment mining in social network is presented to be helpful to many sorts of people who want to know the public sentiment about their product or other thing. We propose a text analysis algorithm which calculates the polarity of textual opinions expressed in social media.This technique focuses sentiment mining in social network through step to step approach.

Keywords: Data Mining; Sentiment Analysis; Social Network; Social Media Marketing; Text Analysis.

I. INTRODUCTION

Nowadays social media's are greatly used by various companies to advertise their product. The aim of these advertising is to introduce their product to the consumers all around the world. This refers to the process of gaining traffic or attention through social media sites. It makes interactions among people in which they create, share, and exchange information and ideas in virtual communities and networks. With the proliferation of web-based e-commerce sites and social media sites, people spread subjective evaluations of products in unprecedented volume and speed.These are full of positive and negative opinions. But there is no appropriate methodology followed by the social networks to analyze the review to find the present status of a marketer's product. So it became hard to collect the feedback for the companies, yet there is an ocean of resource in social media's. From this ocean, mining information is an exceedingly difficult task.

For example, 'Symphony' is the most popular brand in Bangladesh for low price mobile phones. Symphony has a big collection of phone models and brings new models every month. The young people searches different blogs to know about their phones and give opinions. Firstly this system consumes time of the customers to know about a product. Secondly the opinions given by them remain un-combined to justify the phone.

The only hope is the opinion analysis. Sentiment mining, also called opinion analysis, analyzes people's opinions and emotions towards entities such as products, services, organizations, individuals, issues,events, topics, and their attributes.Through opinion analysis it is possible to retrieve appropriate information from the social media sites. Most of the opinions expressed in social media are in textual format. Hence textual analysis is the best approach for sentiment analysis.Text based analysis retrieves the polarity of a textual opinion and approaches to a combined result for large amount of textual data.Our goal is to technical modeling of opinion analysis in social media through the help of artificial intelligence (AI). Through this, the opinions can be divided into separate words which are positive, negative or neutral. To understand the semantic of a total sentence is a natural language processing (NLP) problem. After solving that protecting spamming is a vital issue. Due to solve all of these AI meets with a new term 'psychology' and produces new step to step approaches. These give the way to construct a method to overcome the problem.

The content of the paper discusses firstly, the primary concept about sentiment analysis and some previous works.Next the base topics sentiment analysis is discussed with correlating artificial intelligence. Then the methodology for textual analysis is described with a flow diagram.The applications of the opinion analysis are described shortly with the result discussion. Finally conclusion of the paper, references, and the web address which are used. It also contains the challenging limitations and future works.

II. RELATED WORK

Sentiment analysis or opinion mining refers to the application of natural language processing, computational linguistics, and text analytics to identify and extract subjective information in source materials.

In [1], there are two main approaches to document based sentiment analysis: supervised learning and unsupervised learning. Several systems have been built which attempt to quantify opinion from product reviews. [2] Perform sentiment analysis of movie reviews. Their results show that the machine learning techniques perform better than simple counting methods. They achieve an accuracy of polarity classification of roughly 83%. In [3], they identify which sentences in a review are of subjective character to improve sentiment analysis. The authors [4] identify local sentiment as being more reliable than global document sentiment, since human evaluators often fail to agree on the global sentiment of a document. They focus on identifying the orientation of sentiment expressions and determining the target of these sentiments. In [5], they follow up by employing a feature-term extractor. For a given item, the feature extractor identifies parts or attributes of that item. e.g., battery and lens are features of a camera.In [6],they perform a comparative experiment on sentiment classification for online product reviews. They classifythe online product reviews into positive and negativeclasses and discuss a series of experiments with different machine learning algorithms.

III. Sentiment Mining Approach Based on AI

Here it is needed to automate the system to justify the keywords. Hence it is vastly interrelated to artificial intelligence. For that purpose Artificial Intelligence works with human psychology. The process is done by several AI terms step to step.

A. *Polarity of Lexicons*

Firstly the opinion is divided into opinion words that are commonly used to express positive or negative sentiments. For example, good, wonderful, and amazing are positive sentiment words, and bad, poor, and terrible are negative sentiment words. Such words are called lexicons. But the challenge is a word may have different polarity in different situation. If the system can understand the psychology of the speaker, then and only then it can provide the perfect polarity. Here, polarity refers to good or bad.

B. *NLP Problem*

The NLP refers to natural language processing. The system does not need to fully understand the semantics of each sentence or document but only needs to understand some aspects of it, i.e., positive or negative sentiments and their target entities or topics. In this sense, sentiment analysis offers a great platform for NLP researchers to make tangible progresses on all fronts of NLP with the potential of making a huge practical impact.

C. *Spam Detection*

Detecting non-necessary and spam opinions is a big deal. To reduce time complexity, non-necessary opinions should be avoided. But as social media's are open, it sometimes influences people to create hidden agendas or malicious intentions. So opinion spamming has become a major issue. Apart from individuals who give fake opinions in reviews and forum discussions, there are also commercial companies that are in the business of writing fake reviews and bogus blogs for their clients. It is important to detect such spamming activities to ensure that the opinions on the web are a trusted source of valuable information. Unlike extraction of positive and negative opinions, opinion spam detection is not just a NLP problem as it involves the analysis of people's posting behaviors. It is thus also a data mining problem [7].

IV. Classification of Strategies

The term 'Sentiment mining' is the combination of two words 'sentiment' and 'mining'. The word sentiment directly refers to the human mind or attitude or mental state. Another one 'mining' means retrieving information from existing data. Hence very simply sentiment mining is a process to know ones attitude. The process can be classified in several systems.

The solution of the problem sentiment analysis can be classified in five strategies. These techniques depend upon a document analysis module which is converts the input into various sentiment words using linguistic tools. The output of the system is some annotations. The techniques are

- Document based
- Sentence based
- Aspect based
- Comparative opinion analysis
- Acquiring the sentiment lexicon

A. *Document based*

In supervised technique document based analysis require a finite set of classes by which the document should be classified and training data is available for each class. The classes can be positive, negative neutral etc. or some other scale into which the document should be placed. Unsupervised technique determines the semantic orientation of some target keyword within the document. If the average semantic orientation of these keywords is above some predefined threshold the document is classified as positive and otherwise it is defined negative

Practically if an opinion document evaluates more than one entity, this system fails. It is only valid for one entity and one user problem.

B. *Sentence based*

To overcome the limitations of document based solution, the sentence based solution can be helpful. Instead of taking whole document, it just takes a single sentence. Obviously a single sentence refers to a single opinion by a single customer. The entire document may have many opinions of many customers.so the output gained from the split sentences are summarized finally to achieve the result.

C. *Aspect Based*

If the product has many attributes then the people may have a different opinion about each of the attributes. Aspect based sentiment analysis is a feature-based sentiment analysis that focuses on the recognition of all sentiment expressions within a given document and the aspects to which they refer.

D. *Comperative Sentiment Analysis*

There are many type of people give opinions in social media and their expressions are various.so most of the time the input text do not match with the fixed format, though the semantic meaning of the text is similar to other. The solution is to compare.

E. *Acquiring the Sentiment Lexicon*

The big question is how to define a word positive or negative. There are already some approaches for that purpose. Firstly the manual approach. By which the polarity of the words are defined manually. This is not feasible for a huge amount of data. The second one is keeping some seed words [8] related to the document which can define the polarity of a word automatically.

V. METHODOLOGY

A. *System Design*

To construct a model for a system, using the sentiment analysis strategies into a social network can be discussed in two steps [9]. The steps are:

- Average rating.
- Textual analysis.

In this this paper only textual analysis is described. Before that we need to design the variables. For a given collection of products P= { ,.... ,…, },and assume each product , is reviewed by customers. Therefore each is associated with:

- A set of opinions, $O_i = \{o_i^1, \ldots, o_i^j, \ldots, o_i^{C_i}\}$, where each o_i is a piece of evaluative text reflecting a consumer's opinions on the given product, which can be null if the consumer chooses not to provide it.

B. *Text Analysis:Proposed Algorithm for Step to Step Approach*

The process of sentiment analysis has step to step activities [10]. Each step prepares result for the next step. Fig.1 shows the entire system based on following algorithm.

1) Collecting textual data or information from targeted sites or media.

2) Cleaning the extracting data before the analysis is performed. Identifying and eliminating non textual and unnecessary content from the textual dataset is performed here.

3) Appraising and extracting reviews and opinions from the textual dataset through the use of computational tasks.Every single words of a single opinion is considered seperately.

4) Classifying every single words of the opinions as good, bad, neutral etc.

5) Representing the detected polarities considering the need of investigators after passing into required equations.

A word can be objective or subjective; if it is subjective, it can carry a positive or negative polarity, with certain intensity .For example, "fantastic" is a strong positive word, and "questionable" is a moderate negative word. Certainly there are a lot of words that are neutral in terms of polarity, e.g., " white. " Formally, given a word w, it is associated with two scores, a positivity score pscore and a negativity score nscore, such that 0⩽pscore, nscore⩽1 and 0⩽pscore + nscore⩽1. If both scores are 0, the word is a neutral one.

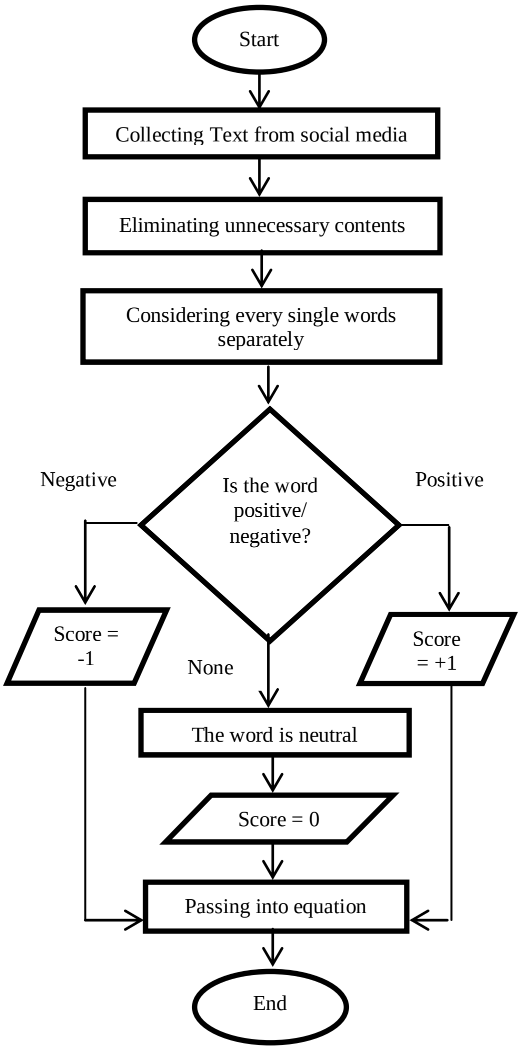

Figure 1. Text analysis process step to step

An excellent lexicon, SentiWordNet, encodes such linguistic knowledge. A data mapping process is illustrated in the table1 through a simple example sentence. Notice that in the PSV, each word is accompanied by its pscore or nscore, depending on its polarity.From Table 1,considering polarity of a word, the score can be assigned by the following criteria:

- If the word is neutral (pscore=nscore=0) or almost neutral (|pscore-nscore|<0.1), then score= 0.
- If the word is positive (pscore-nscore⩾0.1), then score= +1.
- If the word is negative (nscore-pscore⩾0.1), then score=-1.

Considering the above analysis, we come to an equation to find a score value .The proposedequations are,

P (net Pos or net Neg) =P (Pos)-P (Neg) (1)

$$\text{Net Score} = \frac{\sum_{k=0}^{N} P(\text{net Pos or}}{N} \quad (2)$$

)

Here,

- P (pos) = Probability of positive polarity.
- P (neg) = Probability of negative polarity.
- P (net Pos or net Neg) = Net probability of positive or negative polarity.
- Net Score = Final result for making decision.

VI. RESULT DISCUSSION

A data mapping process is illustrated in the Table 1 through a simple example sentence. Notice that in the PSV, each word is accompanied by its pscore or nscore, depending on its polarity.

Table 1. Text Analysis Table

Words	Polarity Score Vector(PSV)	Individual Score
The	Neutral	0
Symphony	Neutral	0
Mobile	Neutral	0
Outlook	Positive	1
Is	Neutral	0
Smart	Positive	1
,	Neutral	0
Its	Neutral	0
Sound	Positive	1
Quality	Positive	1
Is	Neutral	0
Good	Positive	1
,	Neutral	0
But	Negative	-1
Camera	Neutral	0
Is	Neutral	0
Poor	Negative	-1

In the Table 1, we have got P ('1') =5/17 and P ('-1') =2/17.Similarly after getting P ('1') and P ('-1') from all the opinion texts, Eq. (1) and Eq. (2), is used to calculate final result. Hence the sentiment in socialnetwork for anything can be retrieved by the above methodology.

Time can be included in the analysis. Usually, this is graphically displayed through constructing a sentiment time line by plotting the value of the chosen statistic (example frequency, percentages, and averages) over time. The scenario of good-will in the market of a company can be found by the system. Here we have the twitter analysis of apple Inc. at mentioned date from Fig. 2 [11].It describes hourly conditions of the company.

Figure 2. apple Inc. sentiment analysis of a single day.

A review-oriented search engine would have could also serve very well as the basis for the creation and automated upkeep of review- and opinion-aggregation websites. Companies can use this technique for feasibility analysis before taking a project. For political cases it can be used to make an unauthorized survey to check the popularity. Previously it is used in several political issues to know public sentiment. It has specifically been proposed as a key enabling technology in rule making, allowing the automatic analysis of the opinions that people submit about pending organizational policy or government-regulation Proposals.

The general purpose of the analysis is to convert unstructured fragmented text into meaningful information [12]. Once the analysis is completed, a number of conventional options are used to display the result of text analysis. Chief among them is the use of graphical displays such as pie charts, bar charts and line graphs. The polarity is segmented on color, frequencies, percentages and size.

VII. CONCLUSION

In conclusion, sentiment analysis is a relatively new in the context of research. Nevertheless, the contribution to real time conversion of mass volume of textual data into meaningful information can be very useful, especially for social media marketing. The job is to finding out the 'pearl' from the ocean. Ocean refers to the amount of distinct opinions building in social media and pearl refers to the desired information. If marketers can make it possible, then the 'pearl' will give us a golden era of social media marketing.The body language of a human is the 50% of the sentiment. But here is no way to measure that. These are the challenging limitations for future. The detection of spam opinions are still a great challenge.

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REFERENCES

[1] (2013, April) The ACM website. [Online]. Available: http://cacm.acm.org/magazines/2013/4/162501-techniques-and-applications-for-sentiment-analysis/fulltext.

[2] Bo Pang, Lillian Lee, and Shivakumar Vaithyanathan, “Thumbs up? Sentiment Classification using Machine Learning Techniques”, In Proceedings of the Conference on Empirical Methods in Natural Language Processing (EMNLP '02), Philadelphia, pp. 79-86, July 2002.

[3] Bo Pang and Lillian Lee, “A Sentimental Education: Sentiment Analysis Using Subjectivity Summarization Based on Minimum Cuts”, In Proceedings of the 42nd Annual Meeting on Association for Computational Linguistics (ACL '04), pp. 271-278, 2004.

[4] Tetsuya Nasukawa and Jeonghee Yi,” Sentiment Analysis: Capturing Favorability Using Natural Language Processing ”, In Proceedings of the 2nd International Conference on Knowledge Capture (K-CAP '03), ACM, NY, USA, pp.70-77, 2003.

[5] Jeonghee Yi, Tetsuya Nasukawa, Razvan Bunescu and Wayne Niblack, “Sentiment Analyzer: Extracting Sentiments about a Given Topic using Natural Language Processing Techniques ”, In Proceedings of the Third IEEE International Conference on Data Mining (ICDM '03),*IEEE Computer Society*, Washington, DC, USA, 423-434, 2003.

[6] Hang Cui, Vibhu Mittal and Mayur Datar,” Comparative Experiments on Sentiment Classification for Online Product Reviews”, In Proceedings of the 21st national conference on Artificial intelligence (AAAI '06), AAAI Press, Boston, USA, pp. 1265-1270, 2006.

[7] Katina Michael and Keith W. Miller, “Big Data: New Opportunities and New Challenges”, *IEEE Computer Society*, Vol. 46, No. 6, pp. 22-24, 2013.

[8] Bo Pang and Lillian Lee, “Opinion mining and sentiment analysis,” Foundations and Trends in Information Retrieval, vol. 2, no 1-2, pp. 1–135, 2008.

[9] Zhu Zhang and Xin Li, “Controversy is Marketing: Mining Sentiments in Social Media”,In Proceedings of the 43rd Hawaii International Conference on System Sciences(HICSS '10),*IEEE Computer Society*, Washington, DC, USA, pp.1-10, 2010.

[10] Meena Rambocas, Joao Gama,”Marketing Research:The Role of Sentiment Analysis”, FEP n. 489 , 2013.

[11] (2013, July) The SMA website. [Online]. Available: http://socialmarketanalytics.com.

[12] Erik Cambria, Björn Schuller, Bing Liu, Haixun Wang and Catherine Havasi,” Knowledge-Based Approaches to Concept-Level Sentiment Analysis”,*IEEE Computer Society*, Vol. 28, No. 2, pp. 12-14,2013.

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IPv6 Deployment in Bangladesh: An Analysis of the Present State and theWay Forward

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***Abstract*— Internet Protocol version 6 (IPv6) is a new internet protocol, which provides more advance features than Internet Protocol version 4 (IPv4). The World today, acknowledges the importance and need of transition to IPv6. The need of IPv6 arose mainly because of the exhaustion of IPv4 address blocks. Bangladesh is also entering into the IPv6 transition phase. This paper aims to give an overview of IPv6, IPv6 Deployment and Implementation, Present status of IPv6 implementation in Bangladesh and proposes recommendations for deploying IPv6 in accordance with the present internet infrastructure in Bangladesh.**

Keywords— IPv6; IPv6 in Bangladesh; IPv6 transition in Bangladesh; Dual-Stack Network; Virtual Network on Cisco Simulation Tool.

I. INTRODUCTION

A. *Overview*

Internet Protocol version 4 (IPv4) has been the unique identifier for devices located in the IPv4 Internet. However, as IPv4 addresses are being exhausted, it has become recent concern for many countries like US, Japan, Malaysia, India, and many European Countries [1]. As an alternative of IPv4, Internet Protocol Version 6 (IPv6) was first introduced in Internet standard document RFC 2460, published in December 1998 [2].

B. *Need for IPv6 in Bangladesh*

According to the report of Geoff Huston [3], Chief Scientist at APNIC the IPv4 allocation rate from allocated block of APNIC can increase and as a result IPv4 remaining block will be exhausted. As Bangladesh gets IP allocation from APNIC, Bangladesh should take initiative in IPv6 deployment as a backup plan. This work is carried out as a response to this ITU (International Telecommunication Union) GSR (Global Symposium for Regulators) paper [4] from Latif Ladid.

II. KEY MILESTONES IN BANGLADESH

Bangladesh Government has already developed IPv6 procurement document. Bangladesh has observed World IPv6 Day, 2011 [5]. IPv6 Forum was established on January 6, 2010 to initiate IPv6 implementation in Bangladesh. [6]. Presently in Bangladesh 16 ISPs have IPv6 allocations from APNIC. 6 ISPs have test deployment of IPv6. Bangladesh Internet Exchange Point (BDIX) has become dual stack supporting IPv6 peering for last two years. Also 4 ISPs are peering in IPv6 at BDIX. IPv6 Forum Bangladesh has also started to work on it [7]. Mango, a private sector IIG, is the first transit provider of IPv6 in BD on June 8, 2011 [8].

III. IPV6 FEATURES

IPv6 includes features such as new header format, large address space, stateless and stateful address auto- configuration, Built in Security, Neighboring node detection, extension headers, etc [9].

IV. IPV6 AS THE FUTURE PLATFORM

According to ‘IPv6 Forum’, there is a huge scope to use IPv6 as building blocks for the following technologies [10]. IPv6 will also offer new opportunities for Ubiquitous communication, VOIP/ Multimedia services, Social networks and sensor network applications [10].

V. IPV6-IPV4 COEXISTANCE

As we have already discussed, present advances demand IPv6 enabled services. Also, there can be a wide range of IPv6 services. These IPv6 networks have to communicate with IPv4 networks. If these networks are not connected, Internet will be divided into two parts IPv6 & IPv4. This is not feasible, as the main aim of internet is to ensure connectivity. Therefore, there is a need for Bangladesh to find out procedures so that IPv4 and IPv6 networks are reachable from anywhere. Now the question remains, how can we transit to IPv6? It is clear that, the present network infrastructure is based on IPv4, and it is not feasible to change the whole IPv4 infrastructure to IPv6. Because it will require all the IPv4 services and application running on IPv4 platform to be changed and all the network devices reconfigured. Therefore, it is better to find a way that both IPv4 and IPv6 services coexist in the internet, and to find a way that IPv4 can connect with IPv6 networks. There are mainly three transition strategies which can be applied. They are: Dual Stack Network, Tunnelling and Protocol Transition [11].

VI. Related Works:IPv6 Roadmaps around the World

A. *World Initiatives*

Japan and EU are some one to two years forward than US in terms of IPv6 [12]. Many governments are taking initiatives with national strategies to address and support it both politically and financially [12]. Spending in IPv6 is somewhat a small percentage of ICT spending till now but they prove to be valuable especially in Asia and Europe. There is a significant growth seen in the allocation of IPv6 prefixes by a variety of Regional Internet Registries (RIR) [12]. Many of the e-projects are focusing on complete migrating to IPv6 [12].

IPv6 is one of the 8 key Infrastructure Projects under the Malaysian MyICMS 886 strategy. IPv6 is prioritized among the three technologies, sensors, broadband and IPv6, by the Ministry of Energy Water and Communication (MEWC) under the RMK-9 (2006-2010) [12].

Indian government took a number of initiatives for implementing IPv6 in the country [13]. One of the first initiatives was the formation of the India Ipv6 Task Force to have all the stakeholders, like the Government organizations, service providers, content and application providers, equipment manufacturers, cloud computing/data centers providers, under a common platform to discuss on the adoption of IPv6 [13].

VII. Methodology

A. *IPv6 Connectivity Testing: A scenario of Bangladesh IPv6 Deployment:*

- **Objective:** To find the key need for IPv6 deployment in Bangladesh

Procedure and Tools:

In this test, we have taken into account whether an end system is able to connect to IPv6 server taking in consideration different combination of Operating Systems and ISP's internet connection in Bangladesh. We assume to have an idea about IPv6 connectivity status in Bangladesh, and to find what can be the next milestone for IPv6 roadmap.

Here, mainly several famous types of OS's were chosen

- Windows 7
- Windows XP
- Windows Vista
- Mint 13 (Linux)
- Ubuntu 12.04 (Linux)

The test was run with 4 giant ISP's in Bangladesh, and kept their name anonymous. So we have leveled their name as ISP 1, ISP 2, ISP 3, and ISP 4 and ISP 5. And we have tried to connect with an IPv6 server (ipv6.google.com). Also, the OS was tested with IPv6 testing website "test-ipv6.com". This information is already known that, Windows XP does have IPv6 support in built, and it can be checked by the command "netsh interface ipv6 show interface". However, IPv6 stack can be installed from command prompt.

Command:

>IPv6 install

It was first checked that whether IPv6 is already installed in Win XP, and the result is negative, which means, IPv6 was not installed. Then we have run the command to install. When the command to install IPv6 stack was run, an error was found.

Result:

Failed to complete the action
Error 0×800704b8

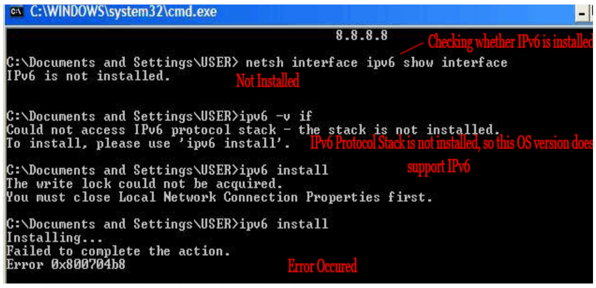

Figure 1. Checking whether IPv6 is installed by default, and Installing IPv6 Stack

This error occurred because Windows security database was not up to date. This problem was solved with the command below, which actually repairs the database.

Command:

>esentutl/p %windir%\security\Database\secedit.sdb

B. *Survey:*

- **Objective:** This survey aims to find out the real scenario of IPv6 deployment in Bangladesh.
- **Target Group:** International Internet Gateway (IIG) Service Providers, Internet Service Provider, and Telecommunication Companies
- **Sample Size**: 15 giant, that is about 500+ employees, and small companies having employees ranging less than 500 basically

taken from Dhaka & Chittagong who have branches all over Bangladesh

- **Analysis Tool:** SPSS

The survey mainly targets to gather information from different ISP/IIGs on how many of their devices are IPv6 capable, whether they have ipv6 address allocation, whether they have taken any initiative to implement IPv6 structure, which kind of routers rule the market, what are the main barriers for IPv6 implementation from the perspective of these organization and in total a overall idea about IPv6 infrastructure and initiatives in Bangladesh.

Sample Population Calculation

This equation below was used for calculating sample size.

$$Ns = \frac{(Np)(p)(1-p)}{(Np-1)(B/C)^{\wedge}2+(p)(1-p)} \qquad [14]$$

- Ns = Completed sample size needed for desired level of precision
- Np = Size of population
- P = Proportion of the population expected to choose one of the two response categories
- B = Acceptable amount of sampling error; 0.17 = ±17 of the true population value
- C = Z statistics associated with the confidence level; 1.645 corresponds to the 90% level

In our case,

Np= 54+36+5+2 = 97 = 100 (approximately) [15, 16]

P= 0.4

B=0.17

C=1.645

Ns = 24/1.30= 18.46 =18 (approximately) = Approximate number of IIG, ISP and Telecommunication Companies proving Internet in Bangladesh. The sample calculations were made keeping in mind that there would be a certain percentage of error.

Procedure:

Representatives of these organizations were contacted through email, and have sent survey URL to them. They were assured that, their names will be kept anonymous. The Survey Questionnaires sample and the screenshot of online format are attached with the index.

C. *Working in Simulation: Creating a Dual Stack Network*

Tool: Cisco Packet Tracer 5.3

Through our survey, it is found out that, technical difficulties matter when it comes to IPv6 transition. So, focus was given to design a simulated Dual Stack "Internetwork" consisting of several networks, to establish that technical difficulties can be overcome with proper tutorial and workshops. It was aimed to apply some of the parameters that are collected during our survey and test its performance. As it was seen in the findings of the survey, Cisco routers rule the market; focus was given on configuring Cisco routers in simulation environment. Cisco simulation tool was used to establish a dual stack network that is connected with an IPv4 server to access information from it. Also a pure IPv4 network was introduced to the dual stack network and it could communicate with the dual stack network.

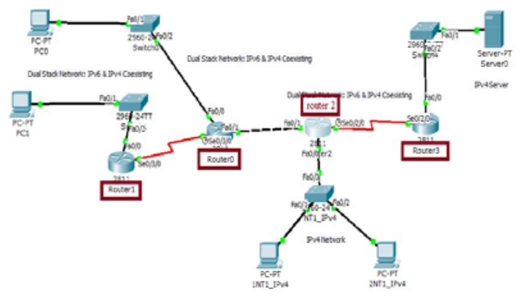

Figure 2. A Dual Stack Network [17, 18, 19]

Right interface of router is configured IPv4 only, and left interface of router 3 is configured both IPv6 and IPv4. In the same way, the bottom part of router 2 is configured IPv4 only and other two interfaces are configured both IPv4 and Ipv6. Router 0 is a pure Dual Stack network.

VIII. Research Findings

A. *Result 1: Connectivity Test Result*

With a simple command we were able to install the IPv6 stack and connect from Windows XP platform to a remote IPv6 server.

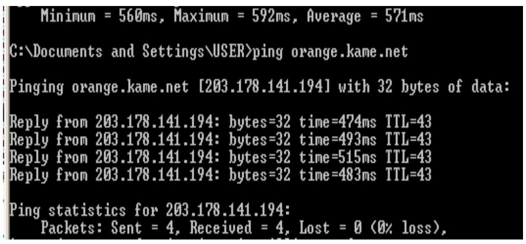

Figure 3. Connecting to IPv6 server through Win XP

From Ubuntu with *ping* command IPv6 servers are not reachable, but with *ping6* it is possible to ping from end system to IPv6 Server. It was not reachable by default in Ubuntu, tunneling software named Miredo had to be used.

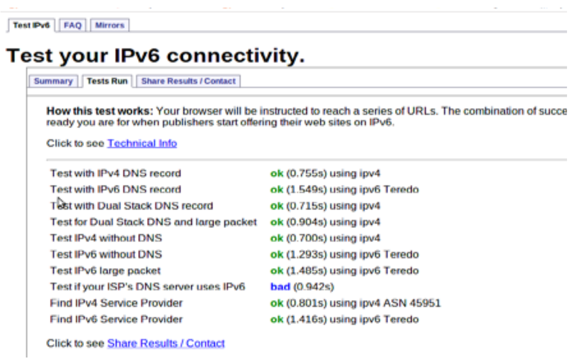

Figure 4. IPv6 Connectivity Test from Ubuntu after Tunneling-1

Figure 5. IPv6 Connectivity Test from Ubuntu after Tunneling-2

Table 1. Summary of Connectivity Test

OS Platform	ISP	Internet Infrastructure	Result
Windows XP	ISP1	Wifi	No
Windows XP – After installing IPv6	ISP 1	Wifi	No
Windows 7	ISP 1	Wifi	No
Windows 7	ISP 2	EDGE	No
Win XP	ISP 2	EDGE	No
Win XP – After Installing IPv6	ISP 2	EDGE	No
Win 7	ISP 3	Wimax	Yes
Win XP	ISP3	Wimax	No
Win XP – After Installing IPv6	ISP 3	Wimax	Yes
Vista	ISP 3	Wimax	Yes
Mint 13	ISP 3	Wimax	No
Win 7	ISP 4	Wimax	Yes
Ubuntu	ISP3	Wimax	No
Ubuntu – With Tunnel Software	ISP3	Wimax	Yes
Android 2.3	ISP4	EDGE	No
Android 2.3	ISP5	EDGE	No

Here the data set can be considered in a two-tuple, {OS, ISP} = Y/N

So we can see,

{Win 7, Var3} = Y,

{Win XP, ISP3} = N,

{Win XP- with IPv6 Enabled, ISP3} = Y

Win 7 supports IPv6 in built. ISP3 is a common factor, so it can be said, as ISP3 supports IPv6, so the end device is able to connect to IPv6 network.

Again,

{Win7, ISP1} = N

{Win XP, ISP1} = N

{Win XP- with IPv6 Enabled, ISP1} = N

Here, ISP1 is common. It seems, if ISP does not support IPv6 packet transfer, then end device is not able to connect to distant IPv6 server. Again, the tuples {Win7, ISP2} = N, {Win XP, ISP2} = N & {Win XP- with IPv6 Enabled, ISP2} = N simply indicates ISP2 does not support IPv6. Therefore we can conclude, even though the most of the OS's are supporting IPv6 if the ISP's don't have IPv6 deployment then end system is not able to connect to distant IPv6 server. As mentioned earlier, IIG of Bangladesh has already provided IPv6 transit. So, this problem is already solved, but still if the particular ISP does not deploy IPv6 then internet packet won't be routed to IIG.

B. Result 2: Survey Result

Analysis 1: Out of the 15 companies, 9 companies answered about their status of IPv6 configurable devices in their system structure. It was significantly seen that very small companies (50 or less employee) on average was found to have zero IPv6 configurable devices, approximately 2 for medium (51 to 120 employees) companies and about 180 devices for very small corporate level companies (120 to 500 employees) and 200+ for medium corporate level companies (511 to 1000 employees). However, there was distinctively high standard deviation seen for small corporate level companies as different such companies had quite varying range of configurable devices.

A. Analysis 2:

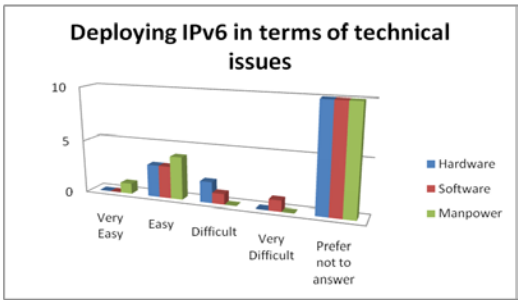

Chart 1. Deploying IPv6 in terms of technical Issues

mc ... dif ... technical issues about all hardware, software and man power implementation. There is substantial difficulty for some in terms of hardware and also some difficulties and very difficulties in software implementation beside a chunk of companies who actually found deploying IPv6 easy in terms of all three hardware, software and manpower. It is very easy in terms of man power for a small chunk of companies as well. The survey revealed with the heterogeneous company that the BD market is overwhelmed by Cisco routers and a small portion is lead by the Mikrotik, Juniper, Huawei and Dell named in decreasing popularity value.

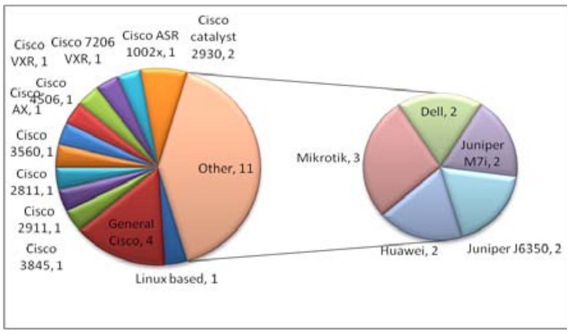

Chart 2: Cisco routers rule the market

C. *Result 3: Simulation Setup*

Successful communication was established with IPv4 only server through a Dual Stack network as shown in Fig. 6.

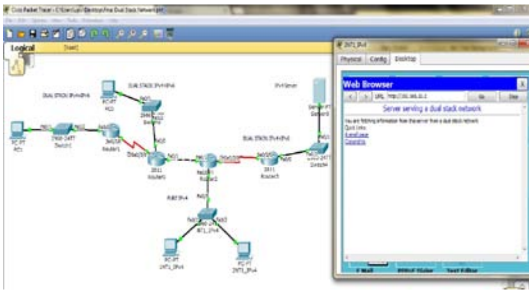

Figure 6 : Browsing IPv4 website from a Dual Stack End System

IX. PROPOSAL

There has to be some objectives set which can be achieved in order to deploy IPv6 all over Bangladesh.

A. *Objective 1: Upgrading the internal Internet Infrastructure*

The internal structure in Bangladesh needs to be put into concentration for getting connected with the IPv6 networks. The findings of this paper illustrates that it is the ISPs who need to deploy IPv6 first in order to provide the end nodes to enter the IPv6 networks as even if the end nodes are configured for IPv6 they would not be able to enter the IPv6 network if the ISPs are not providing this service.

B. *Objective 2: Strategic planning from Corporate Level:*

Japan had a vision of e-Japan which aims to incorporate IT in every sphere of national sectors such as education, health, Industry, Arts & Science, Transportation & Traffic, Social Participation, & Public Administration and so on [20]. Here they aimed to introduce IPv6 based services considering the IP exhaustion that is happening [21].

The country under assessment, Bangladesh, also shows the potential to achieve a significant share in the global IT market. Here, some of the local companies provide software of the world standards of software accreditation certificates like CMMi (Capability Maturity Model Integration) [20]. Moreover, there are more than 150 registered companies that are exporting mobile application solution software and IT –enabled services to different countries [21]. According to news published in New Age [22], Bangladesh is gaining a strong position in global outsourcing market while competing with India and Philippines. Bangladesh outsourcing market is mainly providing IT services to United States, Japan, Denmark and Netherlands. [22, 23]. The fact here is countries like US, Japan, Denmark are going through IPv6 transition phase [24]. So to maintain a sustainable flow of exporting IT services to these countries it is necessary for Bangladeshi Software Company or Corporate to keep pace with the world initiative of IPv6. IPv6 Forum suggests that, if developers start developing IPv6 capable applications, then it can improve the applications with global reach-ability over IPv6 and utilizing mobility support (MIPv6 and NEMO) method, they can start making use of potentials offered by new APIs (Application programming interface) [25]. Therefore, the IT corporate or companies can start researching on enhancing their present services and applications deployed on IPv6.

C. *Objective 3: Encouraging IT Organization/Industries and Professionals*

It was identified that the mind set of many organizations' reluctance to go for the new IP system is one of the barriers in deploying IPv6. In this case, initiatives are required from the government level with awareness programs to encourage these companies. Organizations, Industries and Professionals can also be encouraged by arrangement of technical workshops on IPv6 [26, 27].

X. CONCLUSION

The current scenario of scarcity of space for the internet allocation is the key challenge that the world is facing now. Internet Protocol Version 6 (IPv6) comes as a solution here with a few more benefits of better Quality of Service (QoS), simpler configuration, and higher security at IP level. However, the solution which is a 128 bit address, doesn't aim to remove the existing IP system, the 32 bit IPv4 addresses, but to coexist with it in an effective manner incorporated in the network to ensure world connectivity with networks of both IPv4 and IPv6 types. The paper discusses the IPv6 protocol features for different routing topologies and shows the commonalities and differences with IPv4. It also discusses some techniques for transitioning to IPv6 networks and demonstrates one of the techniques that can be implemented to make IPv4 coexist with the IPv6. The survey of the paper shows that some of the key issues in deploying IPv6 are the business mechanisms and political dimensions that were, however, not faced by IPv4. Thus some vague assumption such as considering it as a costly transition and adequate service gained from the IPv4 that is no different than the native IPv6 makes it tougher and reduces the perception of the value of IPv6. So a lot of government initiatives are expected to make sure that Bangladesh develops in the same rate that our neighboring developed and concerned countries are moving keeping the future networking under consideration for ensuring world connectivity.

REFERENCES

[1] (2013, April). The IPv6 Forum Website. [Online]. Available: http://www.ipv6forum.com/

[2] (2013, April) The IETF website. [Online]. Available: http://tools.ietf.org/html/rfc2460, 1998

[3] (2013, March) The Geoff Huston Report. Website. [Online]. Available: http://www.potaroo.net/tools/ipv4/

[4] Latif Ladid, “The need for more IPv6 Addresses”. Available: http://www.itu.int/en/ITU-D/Conferences/GSR/Documents/PresentationSession_6%20IPv6LatifLadid.pdf

[5] (2013, April) The IPv6 Bangladesh website. [Online]. Available: http://ipv6.org.bd/

[6] (2013, April). The IPv6 Bangladesh website. [Online]. Available: http://ipv6.org.bd/ipv6-and-bangladesh/

[7] (2013, April). The APNIC Conferences website. [Online]. Available: http://meetings.apnic.net/__data/assets/pdf_file/0003/23664/BD-IPv6Bangladesh.pdf

[8] (2013, March). The Mango Teleservices Ltd website. [Online]. Available: http://www.mango.com.bd/press_release.php

[9] (2008, January). The Microsoft Corporation. “Introduction to IP Version 6”, September 2003, January 2008.

[10] (2013, April). The IPv6 Forum Roadmap & Vision website. [Online] Available: http://www.ipv6forum.com/dl/forum/wwc_ipv6forum_roadmap_vision_2010.pdf

[11] (2013, April). The Cisco Website. [Online]. Availabe:http://www.cisco.com/web/strategy/docs/gov/IPV6at_a_glance_c45-625859.pdf, 2010

[12] 30 -34 (2008, June). The Ministry of Energy, Water and Communications (MEWC) website. [Online]. Available: http://www.nav6.org/Home/National%20Strategic%20IPv6%20Roadmap%20[Last%20Updated%2010%20June%202008].pdf

[13] (2013, April). The Government of India, Ministry of Communications & Information Technology Department of Telecommunications website. [Online]. Available: http://www.dot.gov.in/ipv6/2013/Roadmap%20Version-II%20English%20.pdf, March 2013 [April 10, 2013].

[14] (2000) Dillman Mail and Internet Surveys: The Tailored Design Method, NY: John Whiley & Sonc, Inc. (pp. 206-207)

[15] (2013, March) The Internet Service Provider Association Bangladesh Website. [Online]. Available: http://www.ispabd.org/content.php?content.17

[16] (2013, March) The Bangladesh Telecommunication Regulatory Board Website. [Online]. Available: http://www.btrc.gov.bd/jdownloads/Licensing%20Guidelines/iig_licensee0.pdf [March 15, 2013]

[17] The Cisco website. [Online]. Available: http://e-makane.net/misoc/Download/J@IP6-

[18] Lefty Valle-Rosado, Lizzie Narváez-Díaz, Cinhtia González-Segura, VictorChiPech, “Design and Simulation of an IPv6 Network Using Two Transition Mechanisms”. Internet: http://ijcsi.org/papers/IJCSI-9-6-1-60-65.pdf, November 2012 [March 5, 2013]

[19] Ministry of Energy, Water and Communications (MEWC) “NATIONAL STRATEGIC IPv6 ROADMAP”, Internet, http://www.nav6.org/Home/National%20Strategic%20IPv6%20Roadmap%20[Last%20Updated%2010%20June%202008].pdf, 10 June 2008, [accessed date]

[20] (2013, April). The IT Strategy Headquarters Website. [Online]. Available: Internt:http://www.kantei.go.jp/foreign/it/network/0122full_e.html, January 22, 2001

[21] (2013, April). The Financial Express Website. [Online]. Available: http://www.thefinancialexpress-bd.com/index.php?ref=MjBfMDRfMjVfMTNfMV8xXzE2NzQ5OA==

[22] (2013, April). The New Age BD website. [Online]. Available: http://www.newagebd.com/detail.php?date=2012-11-04&nid=28905#.UX6z-bVTCSo

[23] (2013, April). The News Today Website. [Online]. Available: http://www.newstoday.com.bd/index.php?option=details&news_id=46712&date=2011-12-11, December 11, 2012 [April 11, 2013]

[24] (2013, April). The IPv6 Forum Website. [Online]. Available: http://www.ipv6forum.com/

[25] (2013, April). The IPv6 Forum Roadmap & Vision website. [Online]. Available: http://www.ipv6forum.com/dl/forum/wwc_ipv6forum_roadmap_vision_2010.pdf

[26] (2013, March). The CISCO blog. [Online]. Available: http://blogs.cisco.com/sp/cisco-live-2013-a-showcase-for-ipv6-technology/, April 2, 2013

[27] (2013, April) The National Multimedia wesite. [Online]. Avaialable: http://www.nationmultimedia.com/technology/True-Cisco-jointly-promote-new-Internet-protocol-30183636.html

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Shadow Detection and Removal Based on YCbCr Color Space

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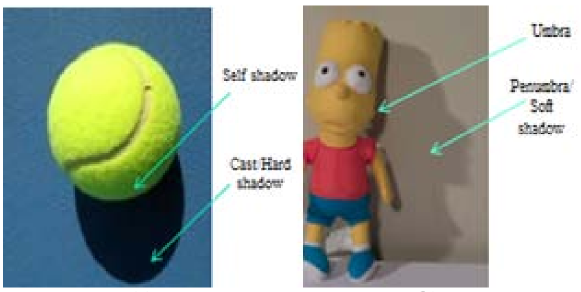
***Abstract*—Shadows in an image may reveal information about the objects shape, orientation and even about the light source. Thus shadow detection and removal is a very crucial and inevitable task of some computer vision algorithms, such as segmentation, object detection and tracking. This paper proposes a simple method to detect and remove shadows from shadow images using YCbCr color space. An approach based on statistics of intensity in YCbCr color space is proposed for detecting shadows. After the shadows are identified, the shadow density model is applied. According to shadow density model, the image is segmented into several regions that have the same density. Then, the shadows are removed by relighting each pixel in YCbCr color space and correcting the color of the shadowed regions in RGB color space. The most salient feature of our proposed method is that after removing shadows, there are no harsh transition between the shadowed parts and non-shadowed parts and all the details in the shadowed regions remain intact.**

Keywords*— *Shadow detection, shadow removal, YCbCr color space.

I. INTRODUCTION

Shadows are physical phenomena observed in most natural scenes. Shadows and shadings in images lead to undesirable problems on image analysis. Moreover, shadows imply the geometric relationship between objects, light source, and viewpoint. This means that real images including shadows are used for image synthesis only in a limited situation where the lighting condition is consistent with that of the real images [16]. That's why much attention was paid on the area of shadow detection and removal over the past decades and covered many specific applications such as traffic surveillance, face recognition and image segmentation. Shadows could be defined as the parts of the scene that is not directly illuminated by a light source due to an obstructing object or objects. The shadow regions, however, are illuminated by ambient light. Typically, shadows can be divided into two major classes: self-shadows and cast shadows as shown in Fig. 1(a). A self-shadow occurs in the portion of an object which is not illuminated by direct light. A cast shadow is the area projected by the object in the direction of direct light. The cast shadow is usually further divided into two parts [15], umbra and penumbra as shown in Fig. 1(b). The umbra represents the shadow region where the primary light source is completely obscured; whereas the penumbra is the region around the edge of a shadow where the light source is only partially obscured [5]. Again based on the intensity, the shadows are of two types: hard and soft shadows as depicted in Fig. 1. The soft shadows retain the texture of the background surface, whereas the hard shadows are too dark and have little texture. Thus the detection of hard shadows is complicated as they may be mistaken as dark objects rather than shadows. Though most of the shadow detection methods need multiple images for camera calibration, the best technique must be able to extract shadows from a single image. This paper proposes a simple method to detect and remove shadows from single shadow images using YCbCr color space.

The paper is organized as follows: In Section 2, related works on shadow detection and removal techniques are reported and briefly described. In Section 3, the proposed shadow detection framework is described and the proposed shadow removal framework is described in section 4. Section 5 presents the experimental results and compares the proposed framework with two other existing methods, outlining the differences and the similarities. Finally, Section 6 outlines the conclusions and the possible future directions of this work.



(a) (b)

Figure 1. Classification of shadows: (a) self and cast shadow and (b) Umbra and Penumbra.

II. PRIVIOUS WORK

It is not difficult for human eyes to distinguish shadows from objects. However, identifying shadows by computer is a challenging research problem. It is therefore of great importance to discover ways of properly detecting shadows and removing them while keeping other details of the original image intact. A lot of research has been performed to detect and remove shadows. Most researches [2, 3, 16] are focused on modeling the differences in color, intensity, and texture of neighboring pixels or regions. Finlayson et al. [17] proposed a method which used illumination-invariant image with the original color image to locate the shadow edges. Those edges were set to zero and the edge representation was reintegrated to get the shadow-free image. This method could work quite well with high-quality images and calibrated sensors, but often performed poorly for typical web-quality consumer photographs. A faster method for shadow removal by averaging the results of reintegration along a few numbers of Hamiltonian paths in the image was proposed in [11]. Fredembach et al. [14] proved that the error propagation during reintegration can be reduced by closing the shadow edges before reintegration. Fredembach et al. [13] suggested that the shadow regions differ from the non-shadow representation by a single constant which could be calculated in a little time. The constant for R, G and B channels were calculated separately. The constant was such that the addition of the shadow region with the constant would reduce the difference between the shadow region and the surroundings. Here, inverse Fourier transforms that were 4 times the size of the image, were required for 2D reintegration and several different Hamiltonian paths were required for 1D reintegration. In [4], the shadow removal had been achieved in three stages. First a 1D shadow-free illumination invariant image was created from which a 2D color representation was derived and then a 3D shadow free color image was generated. Then, the shadow edges were corrected. A region-based approach to detect and remove the shadows from an image was proposed in [6, 16]. The segmented regions in the image were classified based on relative illumination and using a graph cut. Then the labeling of the shadow and non-shadow regions was done and the lighting of shadow pixels was done to recover a shadow free image. Here, initial segmentation was mandatory for shadow detection method.

Xu et al. [12] proposed a method to detect vague shadows in an image using derivatives of the input image. The hard shadows were detected using color invariant image. However, they could not identify soft shadows properly. In this method a shadow-free image was reconstructed by reintegration using Poisson equation. A method to remove the shadows from curved areas retaining the background texture was proposed in [10]. The removal of shadows was achieved by calculating different scale factors for shadow regions and penumbra regions to cancel the effect of shadows. Zhu et al. [8] proposed a method to detect the shadows in single monochromatic image using a shadow invariant, shadow variant and near-black features. They could not remove the shadows. Another method to detect the shadows in a single image using a Tricolor Attenuation Model (TAM) was proposed in [1]. The shadow identification was done followed by generation of an invariant image on which segmentation was performed. TAM is then used to detect the shadows but the dark areas were misclassified as shadows. Salvador et al. [18] proposed a method to identify and classify the shadows in color images. Here luminance and color information were used to detect shadows. In [7], the shadow removal had been done by illuminating the shadow region till it gets the same illumination as the surroundings. Then the texture was retained. They used color and near infrared images for shadow detection and removal. The shadow detection was done using hypothesis test and shadow removal was done using energy function in [2], assuming that the lighting needed in the shadow region is a constant. However, their shadow removal results were not satisfactory. It is seen that most of the works on shadow removal need multiple images and calibrated camera. Methods like reintegration using Poisson equation are time consuming. Also, dark objects are often mistaken as shadows. So, this paper proposes a simple method to detect and remove shadows based on YCbCr color space. Initially, shadow detection is achieved by focusing on Y channel and calculating its intensity. Then shadow removal is achieved by pixel relighting and color correction.

III. SHADOW DETECTION

The shadow detection process could be a primary step for compensation of the shadows and followed by an eventual step of image analysis task, such as object recognition or it could be a fundamental step where the detection results are directly used by 3D shape estimation or similar tasks.

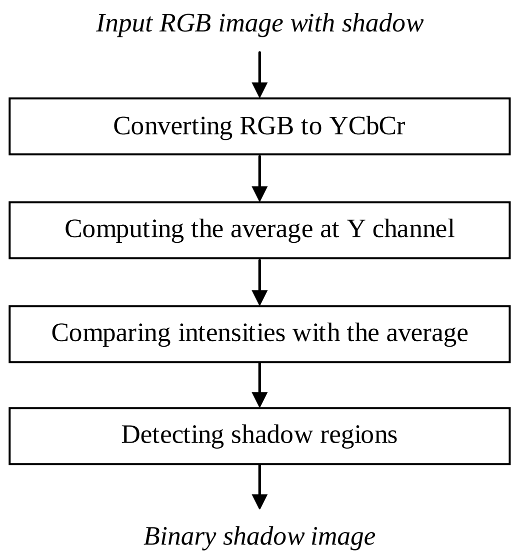
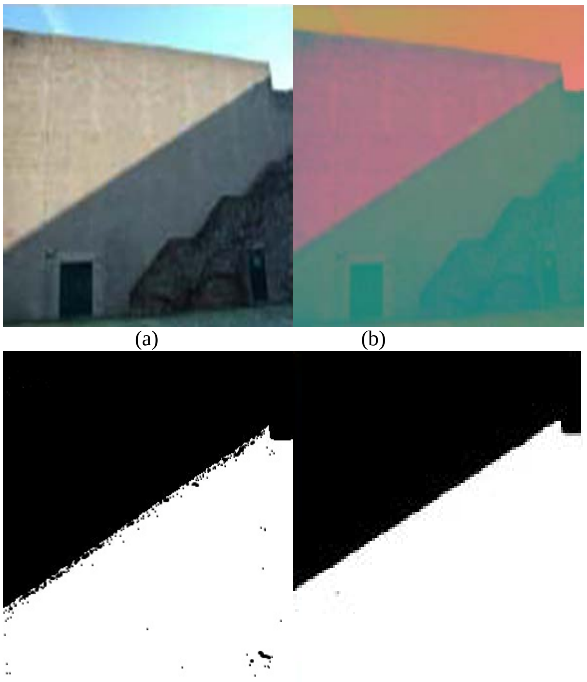


Figure 1. The proposed framework for shadow detection.

In any case, shadow detection is an initial process for a final image analysis task. Hence, the performance of the final task is highly dependent on the shadow detection performance. The proposed framework for shadow detection is depicted in Fig. 2. An approach based on statistics of intensity is presented for shadow detection. Initially the RGB image is converted to equivalent YCbCr image. In YCbCr color space, the Y represents luminance information; Cb and Cr represent the color information.



(a) (b) (c) (d)

Figure 3. Illustration of shadow detection process: (a) input shadow image, (b) converting to YCbCr image, (c) extracted shadow regions after applying first approach and (d) extracted shadow regions after considering non shadow pixels.

Next, focusing on the Y channel its histogram is computed. Histogram dissension gives us a more contrast image at the Y channel. After that the mean of the image at Y channel is computed. Then sliding window iteration through the image is performed. The sliding window size is 3×3. In order to decide which pixels belong to the shadow, two approaches are followed. First, shadow pixels are classified that have the intensity less than one standard deviation of the whole image. This step can't identify the shadow regions properly. As shown in Fig. 3(c), some shadow pixels are identified as non-shadow pixels. So next, the non-shadow point's mean and standard deviation for the sliding window are computed. Now, the pixels that have the intensity less than the one standard deviation of the windows are considered as shadow pixels. Fig. 3 portrays a successful shadow detection process by proposed framework.

Then, isolated pixels are removed using morphological operation. The misclassified pixels are removed using dilation followed by erosion. The result of shadow detection gives us a binary shadow mask which will be used as input of shadow removal process. Some examples of shadow detection using the proposed framework are reported in Fig. 4.

(a) (b)

Figure 4. Examples of shadow detection using the proposed framework: (a) original image and (b) the detected shadow regions marked as white.

IV. SHADOW REMOVAL

The result of the shadow detection is a binary shadow mask, which will be the input to the shadow removal algorithm. In Fig. 5 the proposed framework for shadow removal is shown. For shadow removal a simple shadow model is used, where there are two types of light sources: direct and ambient light. Direct light comes directly from the source, while environment light is from reflections of surrounding surfaces. The shadow model can be represented by following formula:

$$I_i = (t_i \cos\theta_i L_d + L_e) R_i \qquad (1)$$

where, I_i represents the value for the i-th pixel; L_d and L_e represent the intensity of the non-shadow pixels and shadow pixels; R_i is the surface reflectance of that pixel and θ_i is the angle between the direct lighting direction and the surface normal. t_i is the attenuation factor of the direct light; if $t_i = 1$ means the object point is in a sunshine region, if $t_i = 0$ then the object point is in a shadow region. The shadow coefficient for the i-th pixel is denoted by

$$k_i = t_i \cos\theta_i \qquad (2)$$

and the ratio between non-shadow pixels and shadow pixels can be calculated by

$$r = L_d / L_e \qquad (3)$$

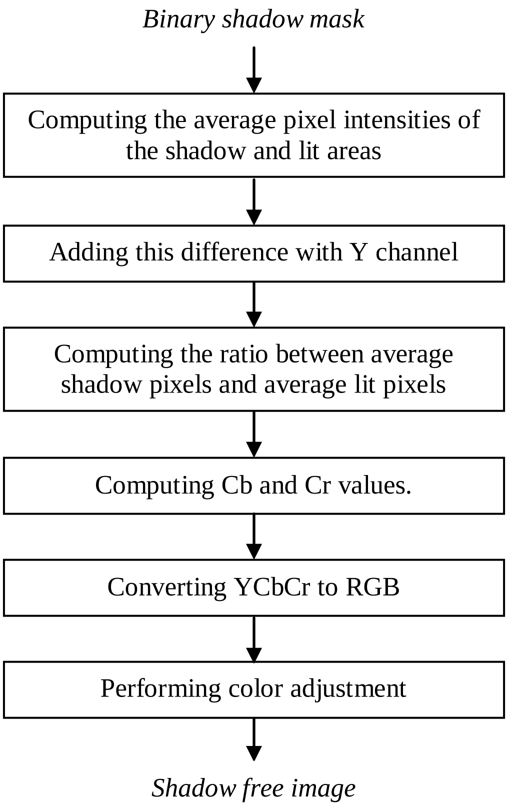


Figure 5. The proposed framework for shadow removal.

The shadow detection procedure provides us with a binary shadow mask where each pixel *i* is assigned a k_i value of either 1 or 0. Based on this model, the goal is to relight each pixel using this coefficient in order to obtain a shadow free image. The new pixel value is computed based on the model proposed in [6]:

$$I_i^{shadow\_free} = ((r + 1) / (k_i r + 1)) I_i \qquad (4)$$

Originally, this model was designed for RGB color space but in this paper the model is modified for YCbCr color space. So, for YCbCr color space R_i and θ_i are not considered. Initially, the average pixel intensities in the shadow and lit areas of the image is computed and added this difference to the pixels in the shadow areas on the Y channel. Then, the ratio between average shadow pixels and average lit pixels is computed. Next, Cb and Cr values are computed. After that the image is converted to RGB image as shown in Fig. 6(c). Because of the ambient light, the ratios of the two pixels are not same in all three color channels. These two pixels will be different not only in intensity, but also in hue and saturation. Thus, correcting just the intensity of the shadowed pixels does not remove the shadow and we need to correct the chromaticity values as well.

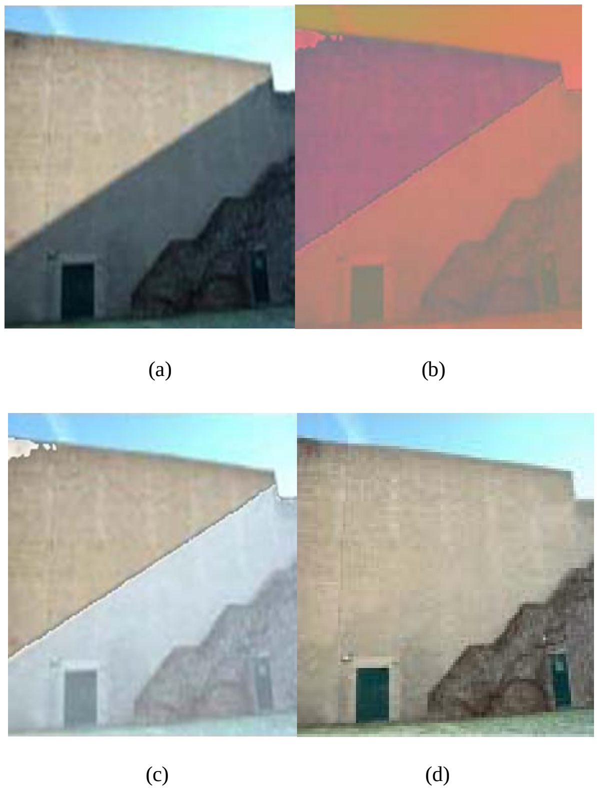


Figure 6. Illustration of shadow removal process: (a) original image, (b) after pixel relighting in YCbCr, (c) after converting to RGB and (d) after color correction and smoothing.

Applying global brightness, the shadow density is calculated which shows the degree of the light's effect [16]. It becomes 0 in a sunshine region, and it becomes 1 in an umbra region. Using the shadow density, the shadow area is segmented into sunshine, penumbra and umbra regions. Since the lighting color of the umbra region is not always the same as that of the sunshine one, the color adjustment is performed between them. Then, the color average and variance of the umbra region are adjusted to be the same as those of the sunshine one. In penumbra, color and brightness adjustments for small regions are performed the same as they are for the umbra region. Finally, all boundaries between shadowed regions and neighboring lit regions are smoothed by convolving them with a Gaussian mask as depicted in Fig. 6(d). Fig. 6 portrays a successful shadow removal process by proposed framework

V. RESULT AND DISCUSSION

The shadow detection and removal module is implemented in MATLAB environment. In the experiments, images of size 256×256 are used. The training set consists of 40 outdoor images and 20 indoor images. Some examples of shadow detection and shadow removal are shown in Fig. 7 and Fig. 8.

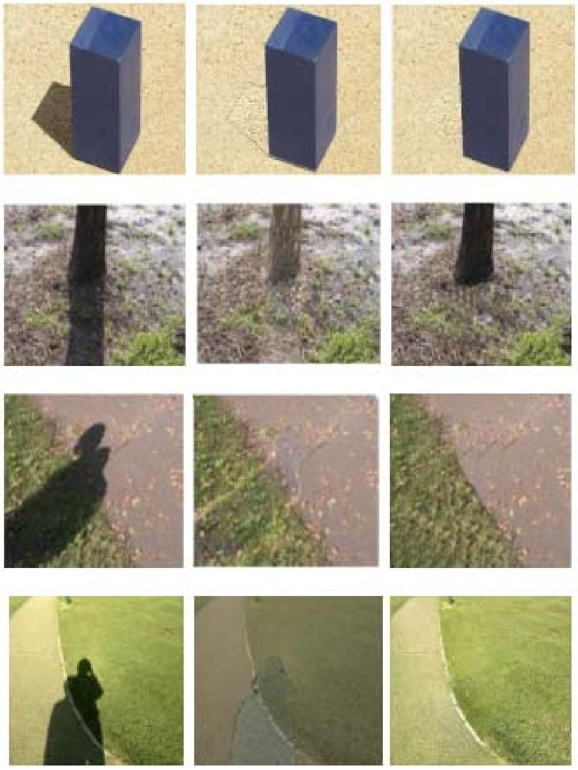
(a) (b)

Figure 7. Examples of shadow detection by using the proposed framework (a) original image (b) detected shadow regions.

(a) (b)

Figure 8. Examples of shadow removal by using the proposed framework (a) original image (b) shadow free image.

Fig. 9 shows the results of applying our framework on some images. The results are compared to the state of the art shadow removal frameworks in [2, 16].



(a) (b) (c)

Figure 9. Comparison of proposed framework with [2, 16]: (a) original image, (b) recovered shadow free image using [2, 16] and (c) recovered shadow free image using the proposed framework.

Column (a) of Fig. 9 shows the input images with shadow and column (b) shows the output of shadow free images using [2, 16], where some shadowed surfaces in the images still do not look similar to the lit parts. Finally, column (c) of Fig. 9 shows the removed shadow regions using proposed framework. It is clearly seen from column (c) of Fig. 9 that the texture of the surface that was under the shadow is preserved and no harsh transition between the shadowed parts and non-shadowed parts. A comparison between the proposed framework and some well-reported methods in the literature is given in Table 1. From Table 1, it can be seen that the proposed framework outperforms the method report in [2] and [16] from the both detection and removal rate point of view.

Table 1. Comparison of detection and removal rates.

Framework	*Shadow detection rate*	*Shadow removal rate*
Kumar et al. [2]	86.2%	82.4%
Baba et al. [16]	82.5%	81.2%
Proposed	91.66%	89.5%

Table 2. Comparison of average computation time.

Framework	*Average Computational time(s) for shadow detection*	*Average Computational time(s) for shadow removal*
Kumar et al. [2]	0.0880	0.9236
Baba et al. [16]	0.0674	0.8209
Proposed	0.0446	0.6207

The average computation time for shadow detection and shadow removal of the proposed framework are 0.0446 and 0.6207 s respectively as shown in Table 2.

VI. CONCLUSION

This paper delineates a shadow detection and removal method based on YCbCr color space. Most of the earlier works involved multiple images along with a calibrated camera whereas the proposed method is a simple and efficient way to remove shadows from single images. Analysis and experimental results suggest that the proposed shadow detection and removal framework is more precise than [2, 16]. Moreover, the proposed framework outperforms [2, 16] because of faster computational time, more precise and easily implementable. In addition, the emphasis of this paper is on the implementation of a new method to detect and remove shadows using YCbCr color space. And emphasis is also given to improve the recovered shadow free image by correcting color. The main achievement of the proposed framework lies in the absence of harsh transition between shadowed parts and non-shadowed parts while keeping all other details intact. It is evident that the proposed framework affectively succeeded in removing shadows from multiple texture images. While conducting the experiments, different viewpoints, illumination conditions and varied distances between object and camera often occurred. We leave these issues for further studies.

REFERENCES

[1] Rani, K.E.E., Jemilda, G., "*Shadow Detection in Single Still Images Using TAM Based Multi Step Algorithm*", IJCST, Vol. 2, Issue 4, 2011.

[2] Kumar, S., Kumar, A., "*Shadow Detection and Removal in Color Images Using Matlab*", International Journal of Engineering Science and Technology, Vol. 2, No 9, pp. 4482-4486, 2010.

[3] Finlayson, G.D., Drew, M.S., Cheng, L., "*Entropy Minimization for Shadow Removal*", Int. J. Comput. Vision, Vol. 85, No 1, pp. 35-57, October 2009.

[4] Finlayson, G.D., Hordley, S.D., Cheng, L.,Drew, M.S., "*On the Removal of Shadows from Images*", PAMI, Vol. 28, pp. 59-68, 2006.

[5] Dare, P.M., "*Shadow Analysis in High-Resolution Satellite Imagery of Urban Areas*", Photogrammetric Engineering and Remote Sensing, Vol. 71, pp. 169-177, 2005.

[6] Guo, R., Dai, Q., Hoiem, D., "*Single-Image Shadow Detection and Removal Using Paired Regions*", In Proceedings of the 18th IEEE CVPR, pp. 2033-2040, 2011.

[7] Germain, A., Salamati, N., Susstrunk, S., "*Removing Shadows from Images Using Color and Near-Infrared*", In Proceedings of IEEE ICIP, pp. 1713-1716, 2011.

[8] Zhu, J., Samuel, K. G. G., Masood, S. Z., Tappen, M. F., "*Learning to Recognize Shadows in Monochromatic Natural Images*", In Proceedings of the IEEE CVPR, pp. 223-230, 2010.

[9] Miyazaki, D., Matsushita, Y., Ikeuchi, K., "*Interactive Shadow Removal from a Single Image Using Hierarchical Graph Cut*", In Proceedings of 9th Asian Conference on Computer Vision, 2010.

[10] Arbel, E., Hel-Or, H. "*Texture-Preserving Shadow Removal in Color Images Containing Curved Surfaces*", In Proceedings of the IEEE CVPR, 2007.

[11] Fredembach, C., Finlayson, G.D., "*Simple Shadow Removal*", In Proceedings of 18th ICPR, Vol. 1, pp. 832-835, 2006.

[12] Xu, L., Qi, F., Jiang, R., "*Shadow Removal from a Single Image*", In Proceedings of 6th International Conference on Intelligent Systems Design and Applications, Vol. 2, pp. 1049-1054, 2006.

[13] Fredembach, C., Finlayson, G.D., "*Hamiltonian Path-Based Shadow Removal*", In Proceedings of 16th British Machine Vision Conference, Vol. 2, pp. 502-511, 2005.

[14] Fredembach, C., Finlayson, G. D., "*Fast Re-Integration of Shadow Free Images*" , In Proceedings of IS&T/SID 12th Color Imaging Conference, pp. 117-122, 2004.

[15] Etemadnia, H., Alsharif M.R., "*Automatic Image Shadow Identification using LPF in Homomorphic Processing System*", In Proceedings of the 7th Digital Image Computing: Techniques and Applications, 2003.

[16] M. Baba and Asada, N., "*Shadow removal from a real picture*," In Proceedings of the SIGGRAPH, 2003.

[17] Finlayson, G.D., Hordley, S.D., Drew, M.S.,"Removing Shadows from Images", In Proceedings of the 7th ECCV, pp. 823-836, 2002.

[18] Salvador, E., Cavallaro, A., Ebrahimi, T., "*Shadow Identification and Classification Using Invariant Color Models*", In Proceedings of the IEEE International Conference on Acoustics, Speech and Signal Processing, pp. 1545 – 1548, Vol. 3, 2001.

[19] Sato, I., Sato, Y., and Ikeuchi, K. "*Illumination distribution from brightness in shadows: Adaptive estimation of illumination distribution with unknown reflectance properties in shadow regions*", In Proceedings of the ICCV Vol. 2, pp. 875–882, 1999.

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Face Detection using RGB Color Model

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***Abstract*—Face is our primary focus of attention for conveying identity. Human face detection by computer systems has become a major field of interest. Detection of faces in a digital image has gained much importance in the last decade, with application in many fields. RGB color model is used in skin color segmentation to separate the human skin pixels. The specific values of red, green and blue components for human skin are described in this research paper. Filtering & labeling of an image and some morphological operations are used to find out face candidate. The different geometrical properties of human face are used to reject the non-human face region. Specific advantages of this approach are that skin color analysis method is simple and powerful, and the system can be used to detect multiple faces. This face detector has been applied to several test images, and satisfactory results have been obtained.**

***Keywords*—Face Detection; Color Based Segmentation; RGB color space; Region Properties.**

I. INTRODUCTION

Detection of the human face is an essential step in many computer vision and biometric applications such as automatic face recognition, video surveillance, human computer interaction (HCI) and large-scale face image retrieval systems. The first step in any of these face processing systems is the detection of the presence and subsequently the position of human faces in an image.

Face detection in color images [1] has also gained much attention in recent years. Color is known to be a useful manner to extract skin regions, and it is only available in color images. This allows easy face localization of potential facial regions without any consideration of its texture and geometrical properties. A Robust Face Detection Algorithm [2] is developed by using Skin Color.

Most techniques up to date are a pixel-based skin detection method [3], which classifies each pixel as skin or “non-skin” individually and independently from its neighbors. Early methods use various statistical color models such as a single Gaussian model [4], Gaussian mixture density model [5], and histogram-based model [6].

A survey of skin color detection can be found in [7]. An automatic face detection system is based on human skin detection, natural properties of faces and the classification strength of Local Binary Patterns (LBPs) and embedded Hidden Markov Models (eHMMs) [7, 8]. A robust multi-face detection system which overcomes several current challenges in the field such as facial expression, occlusion, face rotation, face pose etc. An improved face detection system based on color information, Local Binary Patterns (LBPs) histogram matching, embedded Hidden Markov Models (eHMMs) [7, 8].

A multi-view face detection method uses the edge-based feature vectors [9]. An image based face detection and recognition [10] technique is developed by using Local Binary Patterns (LBPs) and Support Vector Machines (SVMs).

Neural Network method is used in a novel approach for recognition of human face [11]. Artificial Neural Network and Principle Component Analysis are used in Facial Expression Recognition [12].

A new face detection and recognition technique [13] uses skin color modeling and template matching.

Face detection is a challenging task because of variability in scale, presence or absence of structural components, location, facial expression, illumination, orientation (rotated face) and pose (frontal, profile). Those factors are as follows:

Pose: The image of a face varies due to the relative camera-face pose (frontal, 45 degree, profile, and upside down).

Presence or absence of structural components: Facial features such as beards, mustaches, and glasses may or may not be present and there is a great deal of variability among these components including shape color and size.

Facial expression: The appearance of faces is directly affected by a person’s facial expression.

Occlusion: Faces may be partially occluded by other objects. In an image with a group of people, some faces may partially occlude other faces.

Image Conditions: When the images is formed, factors such as lighting (Spectra, source distribution and intensity) and camera characteristics (sensor response, lenses) affect the appearance of a face.

Face size: In the image, human face can exist with different sizes. Sometimes, size of existed face is too small or big to describe clearly facial components information.

The motivation of this research is to develop a face detection algorithm that is very robust against illumination, focus and facial expression. In section II, the proposed face detection algorithm is described.

The proposed algorithm has three primary stages such as, Skin color segmentation, face candidate localization and rejection of non-human face region. RGB color model is used to find out human skin areas from RGB image. Face candidate localizer is described, which is used to find out face candidates. Rejection of non-human face is described by regions properties. Section III represents the experimental results. Finally, conclusions are given in section IV.

II. Our Proposed Algorithm

Our proposed approach has three parts: A) Skin color segmentation, B) Face candidate localization and C) Rejection of non human face skin regions. Fig.1 shows that the flowchart of our proposed approach. Skin Color Segmentation phase uses RGB color space for finding skin pixels from an input image. Selection of RGB color space is the R components is always the strongest one of human skin which has the special expression of blood color. In our analysis, different value of R, G & B components have chosen for the skin pixels.

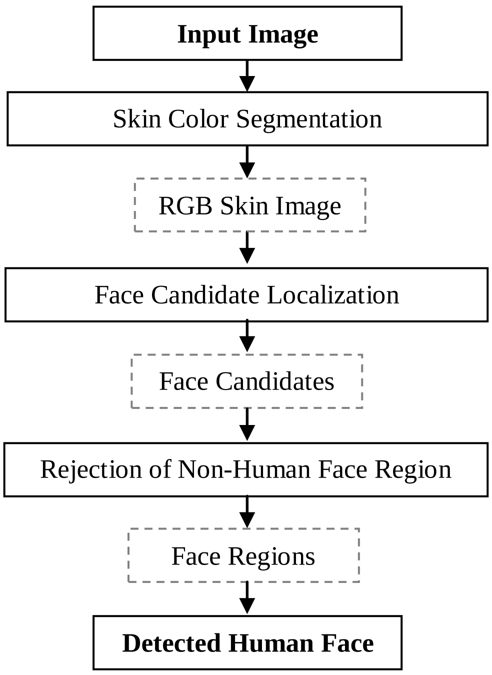


Fig. 1: Face Detection Approach.

Face Candidate Localizer is used to find out most probable face candidate for a human. Morphological operations are used in this term. By applying height and width ratio of each region, face candidates are found. In the 'Rejection of non human face skin regions' part, the most probable face regions are found by using some regions properties such as: area, eccentricity, bounding box, combination of bounding box and area, centriod.

A. *Skin Color Segmentation*

The purpose of skin color segmentation RGB color model is used to find out skin pixels from the input image. Skin color segmentation is shown in fig.2.

The RGB color space consists of the three additive primaries: red, green and blue. Spectral components of these colors combine additively to produce a resultant color.

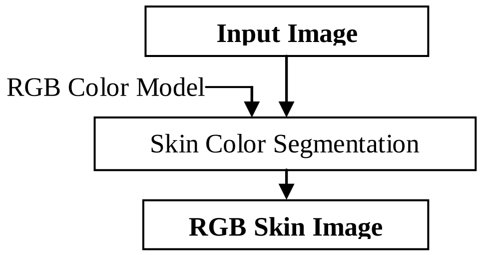


Fig. 2: Skin Color Segmentation

The RGB model is represented by a 3-dimensional cube with red, green and blue at the corners on each axis (Fig.3). Black is at the origin. White is at the opposite end of the cube. The gray scale follows the line from black to white. In a 24-bit color graphics system with 8 bits per color channel, red is (255, 0, 0). On the color cube, it is (1, 0, 0).

The RGB model simplifies the design of computer graphics systems but is not ideal for all applications. The red, green and blue color components are highly correlated.

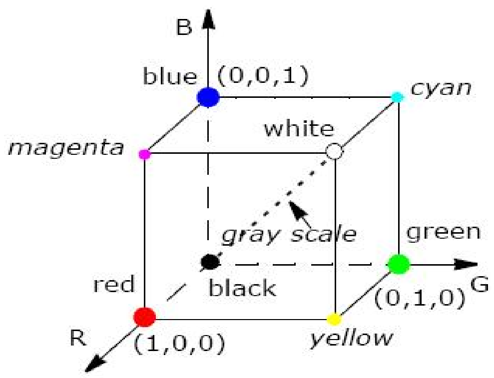


Fig. 3: RGB color space.

Color is a useful piece of information for skin detection. The skin detection is the most common and first approach for detecting meaningful skin color [2], skin color detection may avoid exhaustive search for faces in an entire image. For skin detection task, many color spaces with different properties have been applied. Many researchers have achieved some results with RGB, normalized RGB, HSI, YCbCr and RGB-space ratios.

However, there are many challenges in this task such as different illumination conditions, human faces, and similar skin colors. Our way is to build a skin classifier to define explicitly the boundaries of skin cluster in RGB space which is shown in equation (1).

Decision rules of our skin modeling are as follows:

$$\delta(\mathrm{P}(\mathrm{X},\mathrm{Y})) = \begin{cases} 1; & \textit{if a set of conditions is satisfied} \\ 0; & \textit{Otherwise} \end{cases} \quad \ldots \quad (1)$$

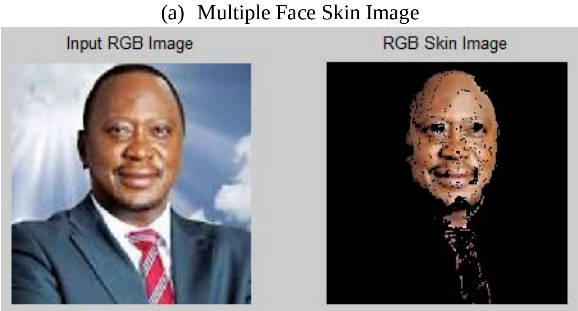
where, *P(x,y)* is a pixel of color image and a set of conditions are listed in Table 1.

Table 1: A Set of Conditions Defining Skin Pixels

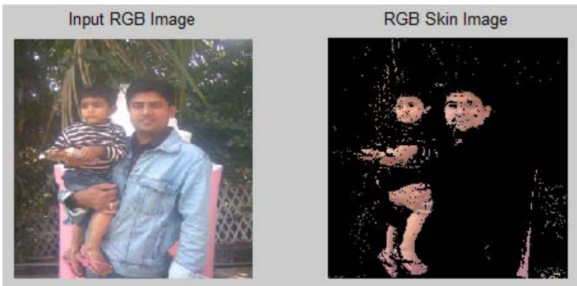
R	R-G	G-B	R-B	(R-G)-(G-B)	B	G
[70,85]	[30,55]	[-5,35]	-	-	[20,255]	[30,255]
[86,100]	[30,60]	[-5,40]	-	-	[30,255]	[40,255]
[101,150]	[0,17]	[-2,10]	[15,75]	-	-	-
	[18,30]	[-255,-10] or [25,45]	-	[-15,285]	-	-
	[31,70]	[-5,90]	[-255,120]	[-20,285]	-	[50,255]
	[71,75]	[-5,0]	[-255,70]	-	-	[50,255]
[151,200]	[15,20]	[0,40]	[20,255]	[-20,285]	-	-
	[21,30]	[-5,0]	[20,255]	[35,285]	-	-
	[31,85]	[-15,70]	[20,255]	[0,285]	[40,255]	[40,255]
[201,255]	[5,25]	[40,70]	-	[-30,285]	-	-
	[26,100]	[0,70]	-	[-15,285]	-	-

The strongest component among R, G & B decides the color. For skin color, generally R component is always the strongest one because human skin has the special expression of blood color. The color is not skin color if the difference between R, G and B are too big or small or R value is smaller than 70. The level of red color affects the decision rule of our skin model. Approximately, the proposed algorithm divides R component into five ranges. The values of R component is must be greater than 70 because, if the value of R component is less than 70 then the color can be dark red, brown, green or black. Few skin detection result shown in fig.4 in various conditions.

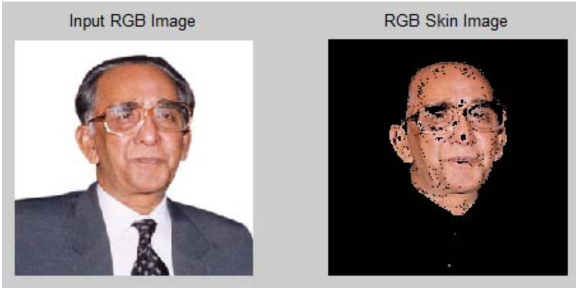
(a) Multiple Face Skin Image



(b) Single Face Skin Image



(c) Multiple Face Skin Image



(d) Single Face Skin Image

Fig. 4: Skin Color Segmentation;

B. *Face Candidate Localization*

Face candidate localizer used to find out the most probable face candidate. The algorithm of this part is shown in fig.5. In the approach of face candidate localization, connected component algorithm is used to label connected skin region. Among those regions, the region which area is smaller than the threshold is rejected by proposed algorithm. In our work, the threshold value is 110 pixels considered as a half of the smallest face size to be detected. This step is called 'Reducing Small Region by Filtering'.

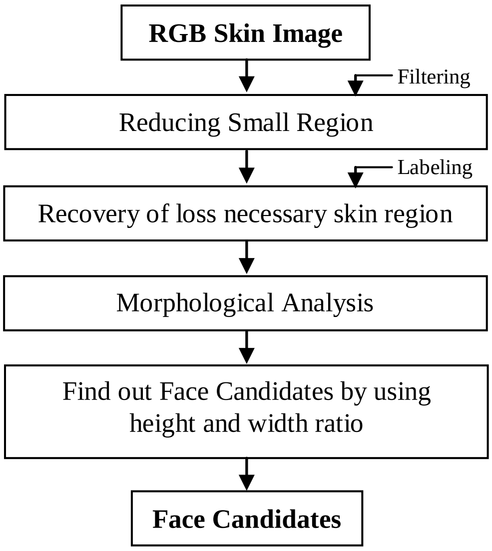


Fig. 5: Face Candidate Localization Method.

Generally skin segmentation is affected by different lighting condition; some real face regions may be lost. To recovery of these necessary regions labeling operation is used to connect non-skin region in each skin regions and change them to skin ones. This process ignores non-skin regions connecting directly to boundaries of their skin regions.

The face candidate localization system involves the use of morphological operations in the next step to refine the skin regions extracted from the segmentation step. Firstly, fragmented sub-regions can be easily grouped together by applying simple dilation on the large regions. Hole and gaps within each region can also be closed by a flood fill operation.

The regions properties – box ratio are used to examine and classify the shape of each skin region. The box ratio property is simply defined as the width to height ratio of the region bounding box. Height/width of a human face is not greater than two times of width/height. Proposed algorithm rejects those regions which are not satisfied by equation (2).

$$\delta(H,W)=\begin{cases}1; & if\,(H<W<2*H) or (W<H<2*W)\\ 0; & Otherwise\end{cases} \quad \ldots \quad (2)$$

where, H is the height and W is the width of particular region.

C. *Rejection of Non-Human Face Region*

To find out the most probable face region which contains human face only, a new algorithm is proposed which used regions properties. A binary image of candidate regions is the most probable human face region. Each face candidates may be real face region if it satisfies some regions properties of human face.

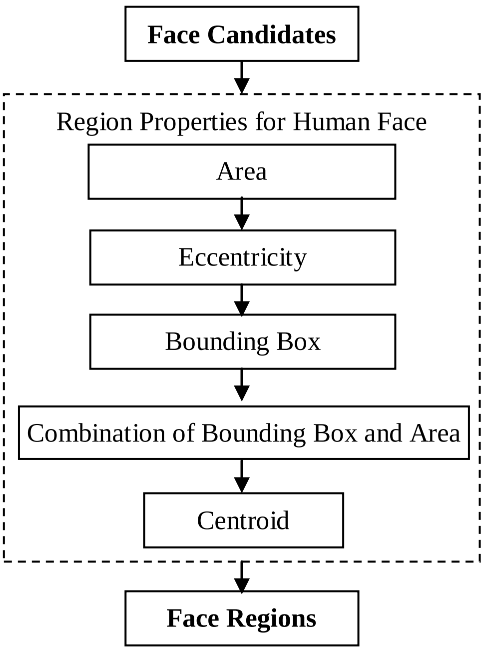


Fig. 6: Rejection of Non-Human Face Region.

These are area (total pixel of particular region), bounding box properties, centroid, eccentricity like oval estimation and combination of these features. A binary image of face candidate passes through these steps, and then these stages of this step remove non human face like regions by using their properties which are fixed according to the human face. The algorithm of this part is shown in fig.6.

Average area of human skin like regions of binary image is calculated to compare this average area with each skin regions of binary image. If any skin region is less than average value, then that skin region will be rejected. Area of any skin region is calculated by counting number of skin region pixels. This method is helpful for removing small skin regions from a binary image and proceeds to next stage for removing non human face skin like regions.

Generally, shape of human face is likely to oval shape, so the region which has an oval shape these are not rejected by this stage, and those regions whose shape likely to line are rejected. An ellipse (skin region) whose eccentricity is 0 is actually a circle, while an ellipse whose eccentricity is 1 is a line segment. The oval shape of a face can be approximated by an ellipse so eccentricity of all skin connected regions are calculated and to discard all skin regions whose eccentricity greater than by 0.89905 and less than 0.3.

Proposed approach rejects non human face skin region based on height to width ratio [4, 6, and 8]. Generally, height to width ratio of skin regions is measurable factor because it is also big factor for rejecting non human face regions. If height to width ratio of skin region is greater than by threshold value, then this skin region will be rejected. Here, the threshold value for height to width ratio is decided as 1.902.

The proposed approach calculates skin region area bounded by bounding box and also calculates this bounding box area. Skin area is the skin pixels bounded by bounding box. Bounding box area is the multiplication of height and width of bounding box. If two times of skin area is less than bounding box area, then this skin region will be rejected.

Generally, the human faces are evenly distributed in the centre, which means the human faces are not present in the side of images. Center of a face region means the center of the skin pixels. Therefore, the centroid of a face region should be found in a small window. Y-axis average centre is calculated from bounding box properties. The proposed approach rejects these skin regions which are not satisfied by equation (3),

$$\delta(\mathrm{P}(\mathrm{X},\mathrm{Y}))=\begin{cases}1; & if\,(Pc \approx Py)<10\\ 0; & Otherwise\end{cases} \quad \ldots \quad \ldots \quad (3)$$

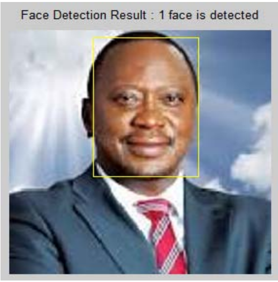
where, $P(x,y)$ is a pixel of binary image, P_c is the center of a skin region and P_y is the Y-axis average center.

III. EXPERIMENTAL RESULT

The proposed face detection method was implemented with MATLAB 7.0. More than 150 images have tested by the proposed algorithm, in which there are many single or multiple faces. These are shown in fig.7.

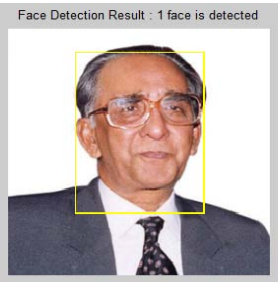
(a) Detection of multiple face

(b) Detection of multiple face

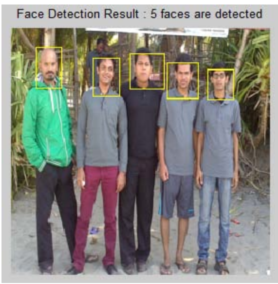


(c) Detection of single face

(d) Detection of single face



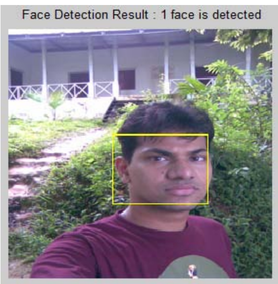
(e) Detection of single face



(f) Detection of multiple face

(g) Detection of single face

(h) Detection of single face



(i) Detection of single face

(j) Detection of single face

(k) Detection of multiple face

(l) Detection of single face

Fig. 7: Detected Human Face;

IV. CONCLUSION

The objective of our research work is to develop an acceptable and efficient face detection algorithm to solve the challenge in face detection technology. In this paper, a strong approach for face detection is applied; the human skin areas are estimated by using skin color segmentation method. RGB color space is used for the color segmentation. Different value of R, G and B had chosen for human skin color. There are many conditions of the difference between R, G and B. After choosing the skin color, some morphological operations are applied and then apply the regions properties such as area, euler number, eccentricity, bounding box, centriod etc. The present algorithm works well for any human face detection.

REFERENCES

[1] Rafael C. Gonzalez, Richard E. Woods, "Digital Image Processing" Second Edition, pp 282-335.

[2] Sanjay Kr. Singh, D.S Chauhan, Mayank Vatsa, Richa Singh, "A Robust Skin Color Based Face Detection Algorithm" Tamkang Journal of Science and Engineering, Vol. 6, No. 4, pp. 227-234 (2003)

[3] V. Vezhnevets, V. Sazonov, and A. Andreeva, "A Survey on Pixel-based Skin Color Detection Techniques", *Proc. Graphicon2003*, Moscow, Russia, September 2003.

[4] J.-C. Terrillon, M. David, and S. Akamatsu, "Automatic Detection of Human Faces in Natural Scene Images by use of a Skin Color Model and of Invariant Moments", *Proc. Int. Conf. AFGR'98*, Nara, Japan, pp. 112-117, 1998.

[5] S.J. McKenna, S. Gong, and Y. Raja, "Modeling Facial Color and Identity with Gaussian Mixtures", *Pattern Recognition*, 31(12), pp. 1883-1892, 1998.

[6] R. Kjeldsen and J. Kender, "Finding Skin in Color Images", *Proc. Int. Conf. AFGR'96*, Killington, Vermont, pp. 312-317, 1996.

[7] Phuong-Trinh Pham-Ngoc, Quang-Linh Huynh " Robust Face Detection Under Challenges of Rotation, Pose and Occlusion" Department of Biomedical Engineering, Faculty of Applied Science, Hochiminh University of Technology, Vietnam.

[8] Phuong-Trinh Pham-Ngoc, Khag-Hyun Jo "Color Based Robust Multi-face Detection under Occlusion, Connection, Attitude and Illumination Changes" SICE-ICCAS, Korea, 2006.

[9] Y. Suzuki and T. Shibata, "An edge-based face detection algorithm robust against illumination, focus, and scale variations," in *Proc. EUSIPCO 2004*, Vienna, Austria, pp. 2279–2282.September 6-10. 2004,

[10] Faizan Ahmed, Aaima Najam and Zeeshan Ahmed, "Image based Face detection and Recognition" International Journal

of Computer Science Issues(IJCSI), Vol. 9, Issue 6, No 1, November 2012.

[11] Dinesh Chandra Jain and V.P. Pawar, "A Novel Approach for Recognition of Human Face Automatically using Neural Network Method " International Journal of Advanced Research in Computer Science and Software Engineering, Vol. 2, Issue 1, January 2012.

[12] Pushpaja V. Saudagare and D.S. Chaudhari, "Facial Expression Recognition using Neural Network", International Journal of Soft Computing and Engineering(IJSCE), ISSN: 2231-2307, Vol. 2, Issue 1, March 2012.

[13] Smt MP. Satone and Dr. G.K. Kharate, "Face detection and recognition in color images", International Journal of Computer Science Issues(IJCSI), Vol. 8, Issue 2, March 2011.

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Intelligent Decision System for Evaluation of Job Offers

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***Abstract*—The word 'Job' term as a regular activity performed in exchange for payment is considered as one of the most important activities for many families worldwide .Evaluation is necessary when more than one opportunity come to an individual personality. Then it requires the job offer evaluation. To fulfill their desired goal, it is the' evaluation' which assesses them well. This involves many factors to be measured and evaluated. These factors are expressed both in objective and subjective ways where as a hierarchical relationship exists among the factors. In addition, it is difficult to measure qualitative factors in a quantitative way, resulting incompleteness in data and hence, uncertainty. Besides it is essential to address the subject of uncertainty by using apt methodology; otherwise, the decision to choose a job will become inapt. Therefore, this paper demonstrates the application of a novel method named Evidential Reasoning (ER) based intelligent decision system(IDS), which is capable of addressing the uncertainty of multi-criterion problem, where there exist factors of both subjective and objective nature. The ER method handles uncertainties by using a belief structure is aggregating degrees of belief from lower level factors to higher level factors.**

Keywords*—Multiple criteria decision analysis (MCDA), uncertainty, evidential reasoning (ER), and IDS

I. INTRODUCTION

When we attempt to evaluate of job offers, it involves multiple criterions such as, location, salary, job content, long-term prospects, safety, and environment, proximity to hospitals, main road, office, transportation cost and utility cost, which are quantitative and qualitative in nature. Numerical data which uses numbers is considered as quantitative data and can be measured with 100% certainty.[4] . On the contrary, qualitative data is descriptive in nature, which defines some concepts or imprecise characteristics or quality of things [5].Hence, this data can't describe a thing with certainty since it lacks the precision and inherits ambiguity, ignorance, vagueness. Consequently, it can be argued that qualitative data involves uncertainty since it is difficult to measure concepts or characteristics or quality of a thing with 100% certainty."Quality of Location" is an example of equivocal term since it is an example of linguistic term. Hence, it is difficult to extract its correct semantics (meaning). However, this can be evaluated using some evaluation grade such as excellent, good, average and bad. Therefore, it can be seen that qualitative criterions which have been considered in selecting a job involves lot of uncertainties and they should be treated with appropriate methodology is Evidential Reasoning Approach (ER), which is a multi-criteria decision analysis (MCDA) method[13][14]. ER deals with problems, consisting of both quantitative and qualitative criteria under various uncertainties such as incomplete information, vagueness, ambiguity [7].The ER approach, developed based on decision theory in particular utility theory [1][11], artificial intelligence in particular the theory of evidence [9][10]. It uses a belief structure to model a judgment with uncertainty. Qualitative attribute such as location or safety needs to be evaluated using some linguistic evaluation grades such as excellent, average, good and bad etc. This requires human judgment for evaluating the attributes based on the mentioned evaluation grades. In this way, the issue of uncertainty can be addressed and more accurate and robust decision can be made. The ER approach has addressed such issue by proposing a belief structure which assigns degree of belief in the various evaluation grades of the attributes.

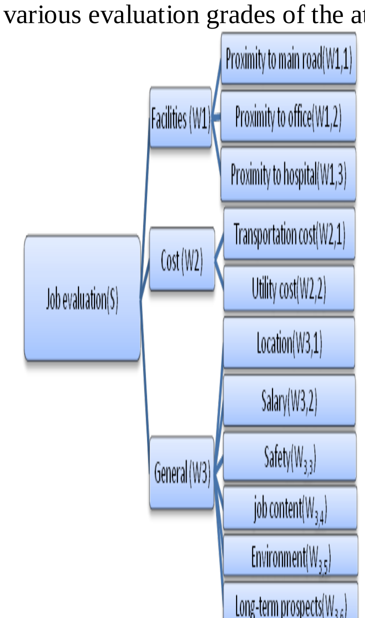

Fig.1. Evaluation Hierarchy for Operation.

In section 2 will briefly represent ER algorithm. Section 3 will demonstrate the application of ER in job evaluation problem. Section 4 will represent the results and achievement. Finally section 5 will conclude the research.

II. EVIDENTIAL REASONING APPROACH

The evidential reasoning algorithm is considered as the kernel of the ER approach. This algorithm has been developed based on an evaluation analysis model [12][13] and the evidence combination rule of the Dempster-Shafer (D-S) theory [9][10], which is well-suited for handling incomplete uncertainty [12]. The ER approach uses a belief structure to model an assessment as a distribution. It differs with other Multi Criteria Decision Making (MCDM) modeling methods in that it employs evidence-based reasoning process to derive a conclusion. The main strength of this approach is that it can handle uncertainties associated with quantitative and qualitative data, related to MCDM problems [13].

The ER approach consists of seven phases[14] including 1) Information acquisition and representation or assessment, 2) weight normalization, 3) basic probability assignment 4) attribute aggregation, 5) Combined degree of belief calculation, 6) utility function 7) ranking

*A. **Assessment***

One of the critical tasks of developing a decision system is to acquire information and to represent them in appropriate format so that it will feed into a model. Since ER approach employs belief structure to acquire knowledge, appropriate information should be selected to feed the ER algorithm, which is used to process the information.

Let 'Job evaluation' (S) be an attribute at level 1 as shown in Fig. 1, which is to be assessed for an alternative (A) (i.e. a job at a certain location) and this assessment can be denoted by A(S). This is to be evaluated based on a set of w_i sub-attributes (such as facilities, cost, general) at level 2, denoted by: $S = \{w_1, w_2, w_3, ...w_i, ...w_n\}$.

Job evaluation (S) can be assessed by using a set of evaluation grades consisting of Excellent(H_1),Good(H_2),Average(H_3),Bad(H_4) and this set can be written as $H = \{H_1, H_2, ...H_n, n = 1,2,......., N\}$. These evaluation grades are mutually exclusive and collectively exhaustive and hence, they form a frame of discernment in D-S terminology.

A degree of belief is associated with each evaluation grade, which is denoted by $\{(H_n, \beta_n), n = 1,......, N\}$ Hence, $A(S) = \{(H_n, \beta_n), n = 1,......, N\}$ denotes that the top attribute S is assessed to grade H_n with the degree of belief β_n. In this assessment, it is required that $\beta_n \geq 0$ and $\sum_{n=1}^{N} \beta_n \leq 1$. If $\sum_{n=1}^{N} \beta_n = 1$ the assessment is said to be complete and if it is less than one then the assessment is considered as incomplete. If $\sum_{n=1}^{N} \beta_n = 0$ then the assessment stands for complete ignorance. In the same way, sub-attribute w_i is assessed to grade H_n with the degree of belief $\beta_{n,i}$ and this assessment can be represented as

$$A(w_i) = \{(H_n, \beta_{n,i}), n = 1,......, N \quad and \quad i = 1,......, n\}$$

Such that $\beta_{n,i} \geq 0$ and $\sum_{n=1}^{N} \beta_n \leq 1$.

The incompleteness as mentioned occurs due to ignorance, meaning that belief degree has not been assigned to any specific evaluation grade and this can be represented using the equation as given below.

$$\beta_H = 1 - \sum_{n=1}^{N} \beta_n \quad .... (1)$$

Where β_H is the belief degree unassigned to any specific grade. If the value of β_H is zero then it can argued that there is an absence of ignorance or incompleteness. If the value of β_H is greater than zero then it can be inferred that there exists ignorance or incompleteness in the assessment. The ER algorithm, as will be discussed, has the procedures to handle such kind of ignorance. It is also necessary to distribute the degree of belief between evaluation grades for certain quantitative input data. For example, sub-attribute 'proximity to hospital', which is at the level 3 of the Fig. 1, consists of four evaluation grades namely Excellent, Good, Average and Bad. When the hospital is located within 1km of the job place, it is considered as excellent, when it is located within 1.5km of the place it is considered as good, when it is located within 2 km of the place it is considered as average and when it is located within 3 km of the place it is considered as bad.

However, when a hospital is located 1.3 km of the place, it can be both excellent and average. However, it is important for us to know, with what degree of belief it is excellent and with what degree of belief it is average. This phenomenon can be calculated with the following formula.

$$\beta_{n,i} = \frac{h_{n+1} - h}{h_{n+1}, i - h_{n,i}}, \beta_{n+1,i} = 1 - \beta_{n,i}$$

$$if \quad h_{n,i} \leq h \leq h_{n+1,i} ....(2)$$

Here, the degree of belief $\beta_{n,i}$ is associated with the evaluation grade 'average' while $\beta_{n+1,i}$ is associated with the upper level evaluation grade i.e. excellent. The value of

h_{n+1} is the value related to excellent, which is considered as 1km i.e. the location of the hospital. The value of h_{n+1} is related to average, which is 1.5 km. Hence, applying equation (2) the distribution of the degree of belief with respect to 1.3 Km of the location of the hospital from the job place can be assessed by using equation (2) and the result is given below:

{(Excellent, 0.4), (Good, 0.6), (Average, 0), (Bad,0)},

B. *Weight Normalization*

In this research Pair wise comparison(AHP) method has been considered for the normalization of the weights of the attribute by considering the following equations

$$.. \ \omega_i = \frac{y_i}{\sum_{i=1}^{j} y_i} \quad ;i= 1\ldots\ldots j\ldots\ldots(3) \qquad \sum_{i=1}^{L} \omega_i = 1 \qquad \ldots\ldots(4)$$

Equation (3) is used to calculate the importance of an attribute (w_i) .This has been calculated by dividing the importance of an attribute (y_i) (this important of the attribute has been determined from survey data) by the summation $\sum_{i=1}^{j} y_i$ of importance of all the attributes. Equation (4) has been used to check whether the summation of the importance of all the attributes is within one i.e whether they are normalized.

C. *Basic Probability Assignment*

The degrees of belief as assigned to the evaluation grades of the attributes need to be transformed into basic probability masses. Basic probability mass measures the belief exactly assigned to the n-th evaluation grade of an attribute. It also represents how strongly the evidence supports n-th evaluation grade (H_n) of the attribute. The transformation can be achieved by combining relative weight (w_i) of the attribute with the degree of belief $(\beta_{n,i})$ associated with n-th evaluation grade of the attribute, which is shown by the following equation.

$$m_{n,i} = m_i(H_n) = w_i \beta_{n,i}(a_l),\,,,$$

$$n = 1,\ldots\ldots, N; \;,,,,,,,,\; i = 1,\ldots\ldots\ldots\ldots, L, \;,,,,,,,,(5)$$

However, in case of hierarchical model, the basic probability mass represents the degree to which the i-th basic attribute supports the hypothesis that the top attribute y is assessed to n-th evaluation grade.

The remaining probability mass unassigned to any individual grade after the ith attribute has been assessed can be given using the following equation.

$$m_{H,i} = m_i(H) = 1 - \sum_{n=1}^{N} m_{n,i} = 1 - w \sum_{n=1}^{N} \beta_{n,i}(a_l),$$

$$i = 1,\ldots\ldots\ldots\ldots, L, \quad \ldots\ldots(6)$$

D. *Kernel of ER approach*

The purpose of ER algorithm is to obtain the combined degree of belief at the top level attribute of a hierarchy based on its bottom level attributes, also known as basic attributes. This is achieved through an effective process of synthesizing/aggregating of the information. A recursive ER algorithm is used to aggregate basic attributes to obtain the combined degree of belief of the top level attribute of a hierarchy, which can be represented as $A(S) = \{(H_n, \beta_n), n = 1,\ldots\ldots, N\}$. In this recursive ER algorithm, all the basic attributes are aggregated recursively in the following manner as shown in Fig. 2.

In this Fig.2 "Facilities" is considered as the top level attribute, which consists of three sub-attributes. The top level attribute "Facilities" can be denoted by w (i) such that i= 1,2,3,..n. This means at this level there could be other attributes. For example, in our case, this level consists of three attributes and the level is considered as second level as shown in Fig. 1. It is interesting to note that top level of Fig.1 contains only one attribute and that can be denoted by So (Job evaluation) and has three sub-attributes at second level. For the top level attribute (S) the combined degree of belief needs to be calculated based on the second level attributes.

From Fig.2 it can be observed that w(1), [considering the value of i as 1] consists of three sub- attributes and hence $w_1 = \{w_{11}, w_{12}, w_{13},\ldots\ldots\ldots\ldots w_{18}\}$ or $w(i) = \{w_{i,j}, w_{i,j+1}, w_{i,J+2},\ldots\ldots\ldots\ldots w_{i,J+n}\}$ such that i=1…..n and j = 1…..L. Taking account of the basic probability assignment and remaining unassigned probability mass of three sub-attributes mass of w_1 matrix (1) has been developed as shown below. These bpa (such as m_{11}, m_{21},,etc and reaming unassigned bpa such M_{H1}) have been calculated by using equations 5 and 6.

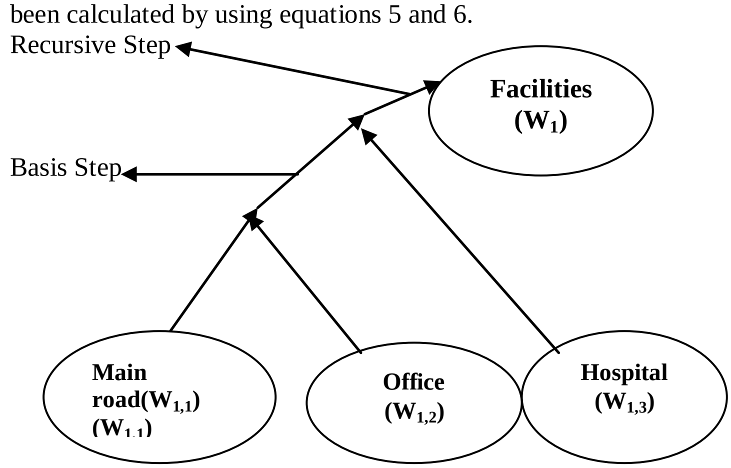

Fig.2. Recursive Manner of Assessment [27]

$$M = \begin{bmatrix} m_{11} & m_{21} & m_{31} & m_{41} & m_{H1} \\ m_{12} & m_{22} & m_{32} & m_{42} & m_{H2} \\ m_{13} & m_{23} & m_{33} & m_{43} & m_{H3} \\ m_{14} & m_{24} & m_{34} & m_{44} & m_{H4} \\ .... & .... & .... & .... & .... \\ .... & .... & .... & .... & .... \\ .... & .... & .... & .... & .... \\ m_{n,I} & m_{n+1,I} & m_{n+2,I} & m_{n+3,I} & m_{H,I} \end{bmatrix} ....(1)$$

$$M = \begin{bmatrix} m_{1I(2)} & m_{2I(2)} & m_{3I(2)} & m_{4I(2)} & m_{HI(2)} \\ m_{13} & m_{23} & m_{33} & m_{43} & m_{H3} \\ m_{14} & m_{24} & m_{34} & m_{44} & m_{H4} \\ m_{15} & m_{25} & m_{35} & m_{45} & m_{H5} \\ .... & .... & .... & .... & .... \\ .... & .... & .... & .... & .... \\ m_{n,I} & m_{n+1,I} & m_{n+2,I} & m_{n+3,I} & m_{H,I} \end{bmatrix} .....(2)$$

From matrix (1), it can be seen that each sub-attribute is associated with five basic probability assignment(bpa), where four first four bpa $(m_{11}, m_{21}, m_{31}, m_{41})$ are associated with four evaluation grades (H_1, H_2, H_3, H_4) and final bpa i.e. $m_{H,i}$ is showing the remaining probability mass unassigned to any individual grades after the assessments on sub-attribute have been considered. Each row in this matrix represents bpa related to one basic attribute or sub-attribute.

Now it is necessary to aggregate the bpa of different sub-attributes. The aggregation is carried out in a recursive way. For example, the bpa of first sub-attribute attribute (which is shown in the first row of the matrix 1) is aggregated with the bpa of second sub-attribute. The result of this aggregation is illustrated in the first row of the matrix (2) and this can be considered as the base case of this recursive procedure since this will be used in the latter aggregation of the sub-attributes. This aggregation can be achieved by using the following equation, which will yield combined bpa (such as $m_{1I(2)}, ...m_{4I(2)}$) as shown in the first row of the second matrix.

$$m_{1I(2)} = K_{I(2)}(m_{11}m_{12} + m_{H1}m_{12} + m_{H2}m_{11}) \quad ....(7)$$

Similarly $m_{2I(2)}, m_{3I(2)}, m_{4I(2)}$ can be calculated.

Where $K_{I(2)}$ is a normalization factor used to resolve the conflict and this can be calculated using the equation (8).

$$K_{I(i+1)} = \left[1 - \sum_{n=1}^{N} \sum_{\substack{t=1 \\ t \neq n}}^{N} m_{n,I(i)} m_{t,i+1} \right]^{-1}, i = 1,........, L-1 ....(8)$$

The aggregation of the third attribute is carried out with the resultant of the aggregation of the bpa of the first two attributes. In this way, the aggregation of the other attributes is carried out and finally, the combined aggregations of all the attributes are obtained. This phenomenon has been depicted in Figure 2, where the combined aggregation is obtained, which will be used to obtain the combined degree of belief for the second level attribute "facilities". Equation (9) represents the more generalized version of equation (7)

$\{H_n\}$:

$$m_{n,I(i+1)} = K_{I(i+1)} \left[m_{n,I(i)} m_{n,i+1} + m_{n,I(i)} m_{H,i+1} + m_{H,I(i)} m_{n,i+1} \right] \quad ....(9)$$

$$m_{H,I(i)} = \overline{m_{H,I(i)}} + \tilde{m}_{H,I(i),} \quad n = 1,............., N, \quad .....(10)$$

$\{H\}$:

$$\tilde{m}_{H,I(i+1)} = K_{I(i+1)} \left[\tilde{m}_{H,I(i)} \tilde{m}_{H,i+1} + \tilde{m}_{H,I(i)} \overline{m}_{H,i+1} + \overline{m}_{H,I(i)} \tilde{m}_{H,i+1} \right] \quad ....(11)$$

$$\{H\}: \overline{m}_{H,I(i+1)} = K_{I(i+1)} \left[\overline{m}_{H,I(i)} \overline{m}_{H,i+1} \right], \quad .... (12)$$

Equation 13 is used to calculate the combined degree of belief by using final combined basic probability assignment, say in this case "facilities".

$$\{H_n\}: \beta_n = \frac{m_{n,I(L)}}{1 - \overline{m}_{H,I(L)}}, n = 1,.........,.N, \quad ....(13)$$

$$\{H\}: \beta_H = \frac{\tilde{m}_{H,I(L)}}{1 - \overline{m}_{H,I(L)}},$$

Where

$$m_{n,I(1)} = m_{n,1} (n = 1,.....N) \quad ....(14)$$

β_n and β_H represent the belief degrees of the aggregated assessment, to which the general factor (such as "facilities") is assessed to the grade H_n and H, respectively. The combined assessment can be denoted by $S(y(a_l)) = \{(H_n, \beta_n(a_l)), n = 1,......, N\}$., . It has been proved that $\sum_{n=1}^{N} \beta_n + \beta_H = 1$

The recursive ER algorithm combines various piece of evidence on a one-by-one basis.

E. *The Utility Function (Ranking Job)*

Utility function is used to determine the ranking of the different alternatives. In this research different job sector have been considered as the alternatives. Therefore, the determination of ranking of the alternatives will help to take a decision to decide the suitable job. There are three different types of utility functions considered in the ER approach namely: minimum utility, maximum utility and average utility. In this function, a number is assigned to an evaluation or assessment grade. The number is assigned by taking account of the preference of the decision maker to a certain evaluation grade. Suppose the utility of an evaluation grade H_n is $u(H_n)$, then the expected utility of the aggregated assessment $S(y(a_l))$ is defined as follows: $u(S(y(a_l))) = \sum_{n=1}^{N} u(H_n)\beta_n(a_l)$

The belief degree $\beta_n(a_l)$ represents the lower bound of the likelihood that a_l is assessed to H_n, whilst the corresponding upper bound of the likelihood is given by $(\beta_n(a_l) + \beta_H(a_l))$ The maximum, minimum and average utilities of a_l can be calculated by:

$$u_{\max}(a_l) = \sum_{n=1}^{N-1} \beta_n(a_l)u(H_n) + (\beta_N(a_l) + \beta_H(a_l))u(H_N)),$$

$$u_{\min}(a_l) = (\beta_1(a_l) + \beta_H(a_l))u(H_1) + \sum_{n=2}^{N} \beta_n(a_l)u(H_n),$$

$$u_{average}(a_l) = \frac{u_{\max}(a_l) + u_{\min}(a_l)}{2}.$$

It is important that if $u(H_1) = 0$, then $u(S(y(a_l))) = u_{\min}(a_l)$ if all the original assessments $S(e_i(a_l))$ in the belief matrix are complete, then $\beta_H(a_l) = 0$ and $u(S(y(a_l))) = u_{\min}(a_l) = u_{\min}(a_l) = u_{average}(a_l)$.

It has to be made clear that the above utilities are only used for characterizing a distributed assessment but not for the aggregation of factors.

III. RESULTS AND DISCUSSION

In the previous section, we have discussed about the ER method and how to implement it. Therefore, in this section we will look at the results from using this method on the different types of job.

FIG.3. shows the assessment distribution which must be done first by employing the transformation equation. Any measurements of quality can be translated to the same set of grades as the top attribute which make it easy for further analysis.

The assessments given by the Decision Maker (DM) in Fig. 1 are fed into IDS and the aggregated results are yielded at the main criteria level (Fig.1)

Attributes	***Acme Manufacturing (A)***	***Bankers Bank (B)***	***Creative Consulting (C)***	***Dynamic Decision Making (D)***
Location	B(0.2)A(0.8)	G(0.4)E(0.6)	G(0.4)E(0.6)	E(1.0)
Job Content	G(0.4)E(0.6)	B(0.2)A(0.8)	B(0.2)A(0.8)	G(0.4)E(0.6)
Safety	B(0.2)E(0.8)	A(1.0)	G(1.0)	A(1.0)
Environment	E(1.0)	G(1.0)	G(0.4)E(0.6)	G(1.0)
Long-term Prospects	G(1.0)	B(0.2)E(0.8)	E(1.0)	B(0.2)A(0.8)
Proximity to Hospitals(Km)	2.3	2.6	2.4	2.0
Proximity to Office(Km)	2.0	1.6	1.0	2.0
Proximity to Main Road(Km)	1.4	1.0	2.1	2.5
Salary(Thousand)	1.0	1.6	1.6	2.0
Transportation Cost(Thousand)	2.3	1.0	1.1	1.4
Utility Cost(Thousand)	2.0	2.3	2.0	2.0

*Fig.3.*Assessment Scores Of Job Sector Based On Sub Criteria
(E-Excellent, G-Good, A-Average, B-Bad)

Alternative	*Excellent*	*Good*	*Average*	*Bad*	*Total DoB*	*Unassigned DoB*
Acme Manufacturing (A)	0.14	0.8	0.04	0.01	0.99	0.01
Bankers Bank (B)	0.16	0.23	0.48	0.13	1.00	0.00
Creative Consulting (C)	0.17	0.70	0.10	0.03	1.00	0.00
Dynamic Decision Making (D)	0.18	0.40	0.40	0.02	1.00	0.00

Fig.4.The Overall Assessment (Alternatives)
(DoB-Degree Of Belief)

Alternative	***Minimum Utility***	***Maximum Utility***	***Average Utility***	***Rank***
Acme Manufacturing (A)	0.850	0.855	0.853	1
Bankers Bank (B)	0.743	0.743	0.743	4
Creative Consulting (C)	0.847	0.847	0.847	2
Dynamic Decision Making (D)	0.808	0.808	0.808	3

*Fig.5.*The Expected Utilities Of Alternative Job

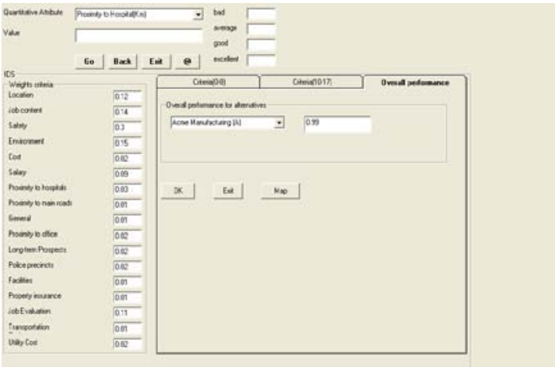

Fig. 6. IDS

The total Degree of belief for each job in FIG.4.does not add up to one, because some of the assessments were incomplete and missing. For example, the total Degree of belief assigned to job alternative is 97%. That is, there is a 3% unassigned degree of belief. The IDS uses the concept of utility interval to characterize the unassigned Degree of belief (or ignorance) which can actually fall into any grade. The ER algorithm generates a utility interval enclosed by two extreme cases where the unassigned Degree of belief goes either to the least preferred grade (minimum utility) or goes to the most preferred grade (maximum utility). The minimum and maximum possible utilities of each alternative generated by the IDS(Figure 6)(based on the given utility values for each grade above) are shown in FIG.5.The job may be ranked based on the average utility but this may be misleading. In order to say that one job theoretically dominates another, the preferred job minimum utility must be equal or greater than the dominated job maximum utility. The ranking of job is as follows:

Acme Manufacturing (A) > **Creative Consulting (C)**> **Dynamic Decision Making (D)** > **Bankers Bank (B)**

IV. CONCLUSION

This paper established the scheme of the application of this evidential reasoning to solve a multiple criteria problem (job offers evaluation) with uncertain, incomplete, imprecise, and missing information. From the results shown above, it is reasonable to say that the evidential reasoning method is a mathematically sound approach towards measuring the job quality as it employs a belief structure to represent an assessment as a distribution. Finally, in a complex assessment as in the job quality appraisal which involved objective and subjective assessments of many basic attributes as shown in Figure 1, it is convenient to have an approach which can deal with the uncertainties or incompleteness in the data gathered. Therefore, the ER is seen as reasonable method for 'quality job' evaluation.

REFERENCES

[1]M Sonmez, G. Graham and **J. B. Yang** and G D Holt, "Applying evidential reasoning to pre-qualifying construction contractors", *Journal of Management in Engineering*, Vol.18, No.3, pp.111-119, 2002.

[2] J. B. Yang, "Rule and utility based evidential reasoning approach for multiple attribute decision analysis under uncertainty", *European Journal of Operational Research*, Vol. 131, No.1, pp.31-61, 2001.

[3]Y. M. Wang, **J. B. Yang** and D. L. Xu, "Environmental Impact Assessment Using the Evidential Reasoning Approach**",** *European Journal of Operational Research*, Vol.174, No.3, pp.1885-1913, 2006.

[4]Lisa M. (2008). *The Sage encyclopedia of qualitative research methods*. Los Angeles, Calif.: Sage Publications. ISBN 1-4129-4163-6.

[5]http://www.pearson.ch/1449/9780273722595/An-Introduction-to-Geographical.aspx

[6]Dodge Y. (2003) *The Oxford Dictionary of Statistical Terms*, OUP. ISBN 0-19-920613-9

[7] D. L. Xu and **J. B. Yang**, "Introduction to multi-criteria decision making and the evidential reasoning approach", *Working Paper Series*,Paper No.: 0106 , ISBN: 1 86115 111 X (http://www.umist.ac.uk/management), Manchester School of Management, UMIST, pp. 1-21, 2001.

[8]A. Taroun and J. B. Yang. "Dempster-Shafer theory of evidence: potential usage for decision making and risk analysis in construction project management." *Journal of the Built and Human Environment Review* 4, no. 1(2011) : 155-166

[9]L. Zadeh, A simple view of the Dempster-Shafer Theory of Evidence and its implication for the rule of combination, The Al Magazine, Vol. 7, No. 2, pp. 85-90, Summer 1986.

[10]Kari Sentz and Scott Ferson (2002); *Combination of Evidence in Dempster–Shafer Theory*, Sandia National Laboratories SAND 2002-0835

[11]Bragge, J.; Korhonen, P., Wallenius, H. and Wallenius, J. (2010). "Bibliometric Analysis of Multiple Criteria Decision Making/Multiattribute Utility Theory". *IXX International MCDM Conference Proceedings, (Eds.) M. Ehrgott, B. Naujoks, T. Stewart, and J. Wallenius,*. Springer, Berlin 634: 259–268.

[12]P. Sen and **J. B. Yang**, "Design decision making based upon multiple attribute evaluation and minimal preference information", *Mathematical and Computer Modeling*, Vol.20, No.3, pp.107-124, 1994.

.[13] B. Yang and P. Sen, "Multiple attribute design evaluation of large engineering products using the evidential reasoning approach", *Journal of Engineering Design*, Vol.8, No.3, pp.211-230, 1997

[14] Tanjim Mahmud and Mohammad Shahadat Hossain,"An Evidential Reasoning-based Decision Support System to Support House Hunting", *International Journal of Computer Applications* 57(21):51-58, November 2012.Published by Foundation of Computer Science, New York, USA..

Analysing Progressive-BKZ Lattice Reduction Algorithm

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***Abstract*— BKZ and its variants are considered as the most efficient lattice reduction algorithms compensating both the quality and runtime. Progressive approach (gradually increasing block size) of this algorithm has been attempted in several works for better performance but actual analysis of this approach has never been reported. In this paper, we plot experimental evidence of its complexity over the direct approach. We see that a considerable time saving can be achieved if we use the output basis of the immediately reduced block as the input basis of the current block (with increased block size) successively. Then, we attempt to find pseudo-collision in SWIFFT hash function and show that a different set of parameters produces a special shape of Gram-Schmidt norms other than the predicted Geometric Series Assumptions (GSA) which the experiment suggests being more efficient.**

Keywords*: *Lattice reduction, BKZ, Gram-Schmidt vectors, SWIFFT.

I. INTRODUCTION

Lattice of our interest is integer or point lattice; discrete subgroup of R^n. It is defined as the integer linear combinations of some vectors (b_1,..., b_d) of the form: $\sum_{i=1}^{d} x_i.b_i$ where xi $\in$ Z. The vectors (b_1,..., b_d) are linearly independent and called a basis of the lattice. The number of vectors d is the lattice dimension.

The goal of lattice reduction is to find good and quality bases that include short enough vectors. SVP or Shortest Vector Problem is the most common and fundamental lattice problem on which many cryptosystems rely on. In lattice-based cryptanalysis one typically look for an approximate variant of SVP, i.e. to obtain a vector $\approx \alpha \times \lambda_1$ (L), where d is the dimension of lattice L, $\alpha \geq 1$ is the approximation factor and λ_1 (L) is the first minimum (the length of the shortest nonzero vector of L). Other important lattice problems are CVP (Closest Vector Problem) and SIVP (Shortest Independent Vector Problem).

Lattice problems are considered as a key element in many areas of computer science as well as cryptography. In particular, public key cryptosystems based on knapsack problem, special setting of RSA and DSA signatures schemes; encryption schemes based on LWE(Learning with error) problems; fully homomorphic cryptosystems are the most important.

Hash function SWIFFT [1], consists of following compression function:

$$\sum_{i=1}^{m} a_i.x_i \in R$$

where $R = Z_q[x]/(x^n+1)$ is the ring of polynomial, $a_i \in R\ (\forall i)$ are m fixed multipliers and $x_i \in R\ (\forall i)$ with coefficients $\in\{0, 1\}$ are the input of length mn. Other parameters of this function include n be a power of 2 and a prime modulus p>0. The algebraic formulation of SWIFFT is equivalent to a lattice of dimension mn and the security of such function depends on the infeasibility of finding relatively short vectors in such a lattice. Finding a collision in this function means to look for a nonzero vector xi ∈ R with coefficients are in {−1, 0, 1} such that $\sum_{i=1}^{m} a_i.x_i$ =0 mod p. In other words, if we can find a vector by reducing the lattice (considering a sublattice of dimension m'n' ≤ mn will be enough) of Euclidean length $\leq \sqrt{mn}$, then it is a *pseudo-collision*.

The most efficient (in terms of reduction quality) BKZ lattice reduction algorithm can able to find such collision in $2^{O(d^2)}$ time complexity. Both theoretically and experimentally we show the variant BKZ-Progressive can do so in reduce amount of time. Precisely, in section 4 we give a heuristic argument that it has enumeration time complexity of nearly $2^{O(d \log d)+O(d)}$ in section 5 we plot experimental results to support this proposition and in section 6 we provide its average runtime complexity for finding pseudo-collision comparing with BKZ. We also showed that it can achieve the quality of reduction as good as BKZ.

II. Related Work

Lattice reduction technique has been studied from the language of quadratic forms (by Hermite in 1850 and then by Korkine-Zolotarev in 1873, together combine HKZ reduction) to analyze current cryptosystems. But it is not seemed to have much improvement since Lenstra-Lenstra-Lovász seminal LLL [2] algorithm has been published in 1982. This polynomial time algorithm is very fast with moderate reduction quality to solve shortest vector problem (SVP). In [3], Schnorr introduced the block concept of HKZ reduction to produce the best reduction algorithm (BKZ) in practice until today. This algorithm gets much slower when block size increases but can achieve approximation ratio (Hermite factor) upto$\approx 1.011^n$ while LLL can achieve roughly upto$\approx 1.022^n$ according to [4]. The practical BKZ algorithm is reported in [5] and has been since widely studied by researchers related to thisfield. Both LLL and BKZ is implemented in NTL package [6] and usually used as a base of any relevant experiments.

Quasi-HKZ reduction technique proposed by Ravi Kannan [7] is another celebrated work that achieve best theoretical bound on enumeration procedure(lattice reduction technique often combines with enumeration routine, for example in BKZ reduction algorithm). An improved analysis of this algorithm is reported in [8] by Stehle´and Hanrot. Recent significant improv ement is done by Gama et al. in [9] where they introduce extreme pruning concept for enumeration. This leads lower success probability to find a short enough vector but gain in overall by saving considerable time. Gama and Nguyen's experiments done in [4] suggest some actual behaviour (output quality v. running time) of lattice reduction algorithms to reduce the gap between theoretical analysis and practical performance of lattice algorithms.

We believe there are two variants of applying BKZ progressively. The first variant uses same lattice basis with increasing block sizes to look for a particular lattice reduction quality. In [10] Buchmann and Lindner used this approach to find sub-lattice collision in SWIFFT, similar approach is used in [11] by Lindner and Peikert to extrapolate BKZ runtime for LWE parameters. The other variant (feed forward approach, i.e. the output of current block is used as the input of next larger block) that used as a preprocessor of the enumeration routine in [12] designated as recursive-BKZ. Our definition of Progressive BKZ is believed to be substantiated the later variant.

III. Preliminaries

A. *Gram-Schmidt Orthogonalization*

The purpose of lattice reduction is to search for orthonormal basis i.e. look for the unit vectors that are pairwise orthogonal in the basis. Gram-Schmidt orthogonalization method can always transfer a basis $\mathbf{B} = (b_1,..., b_d)$ into orthogonal basis $\mathbf{B}^* = (b_1^*, \ldots, b_d^*)$ as follows:

$$b_1^* = b_i$$
$$b_i^* = b_i \sum_{i>j} \mu_{i,j} b_j^*$$

where $\mu_{i,j} = (b_i, b_j)/(b_j^*, b_j^*)$ for all $i \neq j$. We can also define the orthogonal Gram-Schmidt vectors as $b_i^* := \pi_i(b_i)$ for all non-negative i≤d, where π_i denotes the orthogonal projection over the orthogonal supplement of the linear span of $b_1, ..., b_{i-1}$ [see [13] for details]. Every lattice reduction algorithm uses this process to find the reasonably short lattice vectors.

B. *Input and Output basis*

The basis that is chosen for SWIFFT can be represented as the following matrix:

$$\mathbf{B} = \begin{pmatrix} I & 0 \\ H & p.I \end{pmatrix}$$

where the $m * n$ dimensional lattice is symbolized as a $m*n \times m*n$ matrix. H is a $n \times n*(m-1)$ skew circulant matrix in Zp = {0… p−1}; p> 0 is the prime for the SWIFFT parameters. I is the $n*(m-1) \times n*(m-1)$ identity matrix. Right bottom is the n × n scalar matrix and right top is a $n*(m-1) \times n$ dimensional zero matrix.

According to [9], sufficiently random reduced bases except special structure have a typical shape for main algorithms like LLL and BKZ and the shape depends on the ratio $q = \|b_{i+1}^*\| / \|b_i^*\|$ of Gram-Schmidt vectors. This follows Schnorr's *Geometric Series Assumption* (GSA)[14]. The running time of the algorithms depends on q. The BKZ-Progressive also produces same shape as BKZ for certain parameter choice $(m' \leq m)$ of the input basis. As m increases (fixing the n') the $\|b_i^*\|'s$ produce a constant 1 consistently, i.e. the ratio q is 1 too. Seemingly, we can say as long as the basis does not produce special structure the shape is typical. From the discussion of [11] we predict the q reaches its lower bound 1. Now on we will mention the first case as the typical behavior of $\|b_i^*\|$ as well as q and the second case as special behavior. We discuss both cases while analyzing BKZ-Progressive performance and SWIFFT in later sections.

C. *Enumeration:*

Enumeration is a technique that is used alongside lattice reduction for most of the major algorithms. If we imagine a lattice L as a tree then enumeration procedure is simply an exhaustive (depth-first) search for integer combination of the nodes (or vectors) within a upper bound in $\lambda_1(L)$. This procedure (also implemented in NTL) run in exponential time opposed to the lattice reduction technique itself (which run in polynomial time).

IV. THEORETICAL MOTIVATION FOR BKZ-PROGRESSIVE

Block-Korkine-Zolotarev reduction *a.k.a* BKZ reduction is the most successful lattice reduction algorithm in practice. Schnorr and Euchner [5] introduced the following dentition of BKZ reduction.

A. *BKZ Reduction*

Let β be an integer such that 2≤ β<d. A lattice (L) basis $(b_1,..., b_d)$ is β-reduced if it is size reduced and if (for i =1, ...,d−1)

$$\|b_i^*\| \le \lambda_1(\Pi,(L(b_{i,....,}b_{\min(i+\beta-1,d)})))$$

Ravi Kannan provides an algorithm that computes HKZ reduced bases to solve SVP. The main idea of Kannan's algorithm is to spend more time on pre-computing a basis of excellent quality before calling the enumeration procedure. Our BKZ-Progressive approach inspired from Kannan for pre-processing a basis of subsequently increasing block size can be defined as follows.

BKZ-Progressive Reduction: Let BKZ (d, β) denotes the BKZ algorithm running on a lattice dimension d with block size β. Instead of reducing BKZ(d, β) in the first place it will reduce the basis with smaller block size upto β with BKZ(d, 2), BKZ(d, 4), ..., BKZ(d, β).

B. *Heuristic*

A d-dimensional lattice reducing with BKZ-Progressive approach requires enumeration complexity of approximately $2^{O(k \log k)}$, where k is the block size.

Proof. For BKZ reduction with block size k the recent theoretical analysis (see Theorem 2 in [15]) shows that

$$\log q \underset{\sim}{<} \frac{\log(\gamma)}{k-1}$$

where $\gamma \le c_h * k$ is the Hermite constant in dimension k (for constant c_h). The number of BKZ-k iterations sufficient to get this log q is $\underset{\sim}{<}(c' * n^3 / k^2)$ for constant c' . Since progressive BKZ applies for i =2, 4, ..., k − 2 BKZ(i + 2), on an input basis which is the output of BKZ(i), we expect the enumeration time for BKZ(i + 2) to be (from the heuristic estimation described in [13] for solving SVP using enumeration)

$$q^{c(i+2)^2} = 2^{\log(c_h.i)/(i-1).c.(i+2)^2} = 2^{c\log(c_h.i)(i+O(1))}$$

where c is another constant. Since ≤c'* n^3 / $(i + 2)^2$ iterations needed for BKZ (i+2) for i =2, 1,...,,k −2, the total enumeration time (T) for *BKZ-Progressive* would be

$$T \le \sum_i c' \frac{n^2}{(i+2)^2} 2^{c\log(c_h,i)(i+O(i))}$$
$$\le \frac{k}{2} c'.n^3.2^{c\log(c_h,k)(k+O(1))} \simeq 2^{O(k.\log k)}$$

V. EXPERIMENTAL RESULTS

Experiments are done in 2.67 GHz Corei5 (64 bit) Intel processor machine. The BKZ algorithm and associate enumeration routine is the NTL implementations of floating -point version.

A. *Reduction Quality and Shape of* $\|b_i^*\|$

The reduction quality of lattice basis depends on the Gram-Schmidt vector $\|b_i^*\|$, for a good reduction algorithm the value of $\|b_i^*\|$ does not decay too fast. The parameter $q = \|b_{i+1}^*\| / \|b_i^*\|$ is the measure of a lattice reduction quality. According to [13], for LLL reduced basis q ≅ 1:04 (in high dimension) and for BKZ-20 reduction the value is equivalent to 1.025. The slope of the fitted logarithmic linear curve of $\|b_i^*\|$ is also a measure of reduction quality.

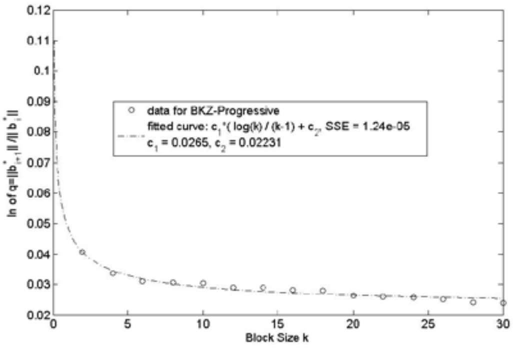


Figure 1. Reduction quality in BKZ-Progressive on an average.

Table 1: Root-*Hermite factor* that can be achieved in BKZ-Progressive reduction with dimension 120.

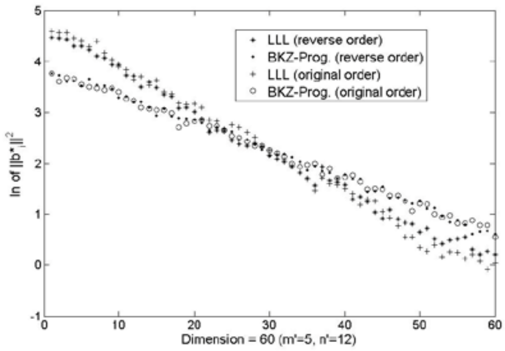
k	*q*	*δ*
2	1.04154	1.02038
4	1.03415	1.01678
20	1.02672	1.01316
22	1.02637	1.01298
24	1.02608	1.01284
26	1.02538	1.01250
28	1.02453	1.01208
30	1.02423	1.01194
32	1.02390	1.01177

In BKZ reduction approach the block size is the parameter that controls the reduction quality. Larger block size gives better reduction in increase of runtime. That means the ratio q should get lower when the block size increases. Experiments done by Gama and Nguyen in [9] (using CJLOSS lattice) found slope = −0.085 for LLL reduction and −0.055 for BKZ-20 reduction (in dimension 110) considering $\left\|b_i^*\right\|^2$ instead of $\left\|b_i^*\right\|$. Figure 1 show in our case, how the log q (the slope) decreases in increase of block size. A fitted curve for th e data points also satisfy that log q $\simeq$ log k/(k−1) (which supports our theoretical assumption for enumeration time).

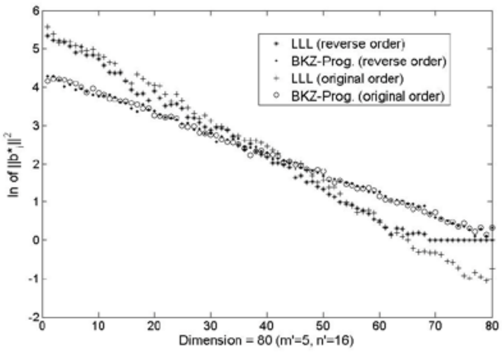
The better reduction directs us to the closer approximation to the shortest vector. The best current *Hermite factor* ($\delta^d = \left\|b_1\right\| / \sqrt[d]{vol(L)}$ is reported in [4] is about 1.0109^d for BKZ-28 reduction. As we know

$$q^{\frac{(d+1)}{2}} = \delta^d$$

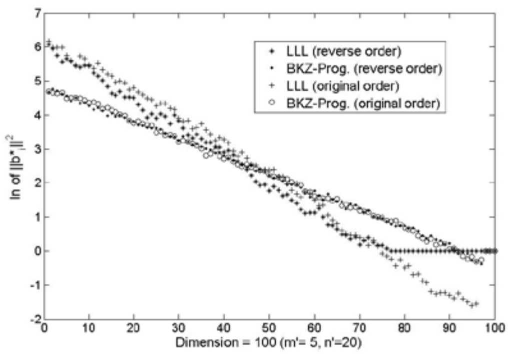
for lattice dimension d, the relation between $q = \left\|b_{i+1}^*\right\| / \left\|b_i^*\right\|$ and *root-Hermite factor* is approximately $\delta \sim \sqrt{q}$. Table 1 shows our experimental outcomes for BKZ-Progressive reduction. The tabulated values are of q, and *root-Hermite factor* for different increasing block sizes(k) of dimension 120.



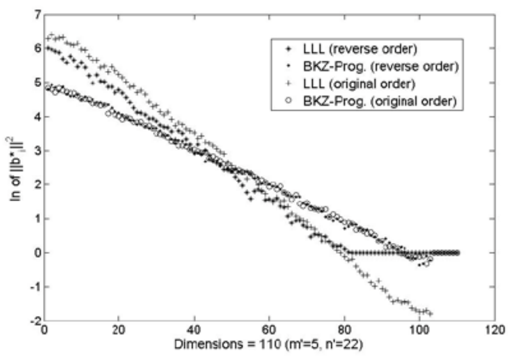
(a)



(b)



(c)



(d)

Figure 2. Shape of $\|b_i^*\|$ curve changes as dimension increases.

It is to be said that the slope of the fitted $\|b_i^*\|$ curve will be independent of lattice dimension. But it is true as long as the Gram-Schmidt norms $\|b_i^*\|$ behaves typically. For our SWIFFT input basis setting we performs experiments to see how the actual shape of $\|b_i^*\|$ looks like after different reduction approach in increase of dimension. We want to see in what dimension the transition (typical to special case) occur and whether the shape of $\|b_i^*\|$ changes differently when we consider change in input basis vector's order. Figure 2 plots this idea. Let us say i' is the dimension after which a transition from typical to special case occurs i.e. $\|b_i^*\|$ (for $i>i'$) becomes constant 1. Also, the current input setting (as in section 3) is considered as original order of basis vectors and a reverse setting of these vectors is considered as reverse order. For original order i' denotes as i' and for reverse order i'_{now}.

For BKZ-Progressive output, the curves of log $\|b_i^*\|$ v.i are about the same for both orders, and are straight line of gradient log (q) for i $\leq i'_{old}$ $\log\|b_i^*\| = 0$ for both orders.

For LLL output, in the reverse order, $\log\|b_i^*\|$ is approximately a straight line with gradient log (q) for $i \leq i'_{new}$, where i'_{new} is the value of line for which the straight line intersects zero. For $i > i'_{new}, \log \|b_i^*\|$ =0 in this order. In case of original order, $\log\|b_i^*\|$ is a straight line of gradient log (q) for i in the interval $i'_{new} > i \leq i'_{old}$ (with same values of log (q) and i' as in reverse order). For $i > i'_{old}, \log\|b_i^*\| = 0$ For $i \leq i'_{new}, \log\|b_i^*\|$ is roughly a convex curve above the straight line of reverse order, intersecting this straight line at $i = 1\, and = i'_{new} + 1 (i.e. \log\|b_1^*\|)$ is approximately same for original and reverse orders and $\log\|b_i^*\|$ =0 for both orders). Also, i' is about the same for BKZ and LLL output (see (c) and (d) of figure 2).

B. ***Enumeration Time Analysis***

We know BKZ reduction procedure implemented in NTL consists of two main parts; enumeration segment which performs exhaustive search in enumeration tree and the reduction segment (based calculate the enumeration time per iteration or block. For progressive approach of BKZ-k reduction, while BKZ-2, BKZ-4,...,BKZ-k reduction is performed subsequently, we consider the last block's enumeration time (i.e. BKZ-k) to compare with that of BKZ only approach. Figure 3 gives a comparison graphs for these two approaches. We consider here dimension 120 with a Schnorr-Hö̈rner pruning [16] parameter 1 to allow larger block sizes.

Least square fitting for the data points of these approaches generate different curve fitting models. In case of BKZ, the fitted model is $f(k)=0.017 * k^2 - 1.1 * k + 16$ with SSE (Sum of Squares due to Error) equivalent to 0.98. On the other hand, BKZ-Progressive fits model $f(k)=0.44 * k * \ln(k) - 1.8 * k + 5.9$ with SSE = 0.25. An alternative model $f(k)=0.062 * k * \ln(k) - 11.25$ also fits quite well with SSE only 0.64.

The above models are generated for special case of $\|b_i^*\|$, for a typical case scenario we get a similar model for BKZ-Progressive approach ($f(k)=0.61 * k * \ln(k) - 2.6 * k + 14$ with SSE = 0.006). A comparison of enumeration time for both cases is listed on the Table 4 in Appendix A.

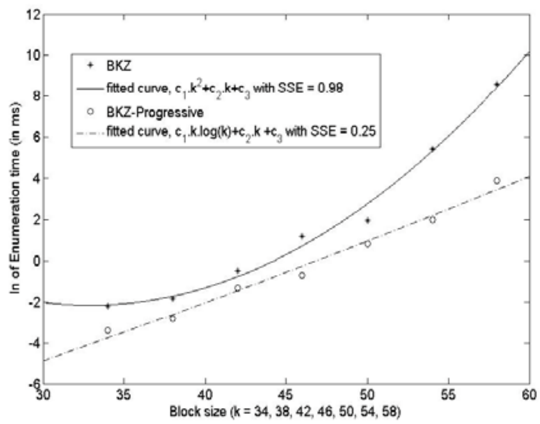


Figure 3. Average enumeration time/block for BKZ and BKZ-Progressive

It shows an improvement over the Peikert-Lindner work in [11] where they used BKZ with block size k directly on LLL reduced basis (default NTL-BKZ implementation), hence resulting in expected BKZ block enumeration time exponential in k^2. We preprocess the basis before applying BKZ-k by progressive BKZ procedure, resulting in a lower on the exhaustive search condition) performs either Trivial, Non Trivial or No Operations reductions (see NTL-BKZ implementation for details). During such a search for the short vector, the enumeration segment is executed number of times. This is the total number of iterations for a BKZ-k reduction. Knowing the total time for those iterations for a block size k, we can expected BKZ block enumeration time exponential in $k * \log(k)$.

VI. CRYPTANALYSIS EXPERIMENTS ON SWIFFT

Again, to obtain *pseudo-collision* we need to find a vector of norm $\|b_1\|$ (usually the first vector in the reduced basis) sssuch that $\|b_1\| \leq \sqrt{mn}$. Following [13] we can calculate the value of $\|b_1\|$ for certain parameters (d,n,p and q as their usual meaning) as follows:

$$\|b_1\| \approx q^{(d-1)/2} vol(L)^{1/d} \approx q^{(d-1)/2} \sqrt[d]{p^n} \qquad (1)$$

Where $vol(L)$ is the volume of lattice equivalent to the determinant of it $(= p^n)$. The usual parameters choice for SWIFFT are *m = 16, n = 64 and p = 257.*

A. *Models for Typical and Special Case*

In typical case the graph of log of $\|b_1\|$ *versus i (for i = 1, ..., d)* is a straight line with gradient *log q* (that depends on the block size *k).*This means that

$$\log\|b_i^*\| = \log\|b_1\| - (i-1) * \log q \qquad (2)$$

Since we know that

$$\sum_{i=1}^{d} \log\|b_i^*\| = \log(vol(L)) \qquad (3)$$

Substituting (1) in (2) gives us

$$\log\|b_1\| = (d-1)/2 * \log + 1/d * \log(vol(L)) \ (4)$$

This is the relation we have in equation (1). From (4) and (2), we can see that the quantity *1/d * log(vol(L))* is approximately equal to $\|b_i^*\|$ *for i ~ (d −1)/2, i.e.* the straight line crosses the value *1/d*log(vol(L))* approximately in the middle. This means the lowest value of the line, log $\|b_1\|$ is about *1/d*log(vol(L)) − (d+1)/2** log q. So, the condition for the typical case should be that $\log\|b_d^*\| \geq 0$.

Then for the special case, $\log\|b_i^*\|$ falls with a gradient *log q for i =1, ..., i' (where i' <d) and* $\log\|b_i^*\|$ *=0 for* $i \geq i'$. Then using (3) we get the following relation in place of (4)

log $\|b_1\|$ *=(i' − 1)/2 * log q +1/i' * log(vol(L))* (5)

Similarly to the above, (5) means that $\log\|b_i^*\|$ *=0= 1/i' ∗ log(vol(L) − (I' − 1)/2 ∗ log q,* which is quadratic equation in *i'* as a function of log q and vol(L). By solving this equation we get a model for *i'*. These have been used to find the *pseudo-collision* in following section.

B. *Pseudo*-Collision *Parameters*

Micciancio and Regev in [17] first observed experimentally that shortest vector of length $l = 2^{2\sqrt{n \log p \log \delta}}$ can be obtained for a optimal dimension $d = \sqrt{n \log p / \log \delta}$. Based on this idea, in our case we can only satisfy the condition of *pseudo-collision* for much smaller n' (n = 20). The other parameters we consider are δ =1.0117 and p = 257 fixed in our case. For larger n, a successful collision is not possible if we restrict these parameters. A smaller choice of parameter p and δ can obviously find collision for larger n (as well as d). In fact, in [10] a smaller choice of modulus p has been actually considered for *pseudo-collision* (see section 6.2) estimate.

We certainly cannot decrease the value of δ as it is reported from [4] as optimum for current best algorithms and also our experiments (in table 1) support this. However, it

Table 2: Choice of parameters for finding *pseudo-collision* in special case.

D	m'	n'	P	k	q	$\|b_1\|$	$\|b_1\|'$
96	6	16	257	4	1.03415	11.54	9.70
108	6	18	257	10	1.03081	11.60	10.10
120	6	20	257	12	1.02946	12.52	10.58
140	7	20	257	14	1.02934	12.53	11.57
151	7	22	257	18	1.02811	13.55	12.28
168	7	24	257	22	1.02637	13.89	12.85
175	7	25	257	21	1.02608	11.19	13.22
182	7	26	257	28	1.02538	15.60	12.20

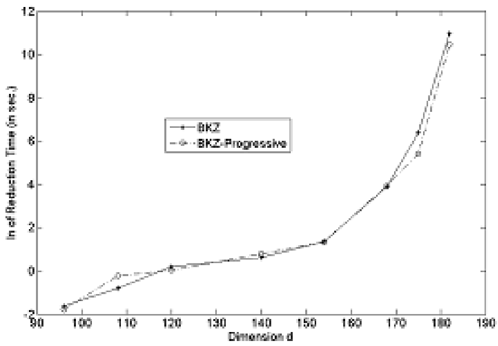


Figure 4. Average runtime for *pseudo*-collision.

would be somewhat different if we considered the special case shape of $\|b_i^*\|$ and extract gradient *log q* from this shape instead of straight line model. But a close investigation reveals that a situation like special case happens when the input vectors are smaller in length (i.e. for larger m and smaller *n'*) in the basis.

Experimental estimate of *pseudo-collision* is listed in table 2 where eight different instances can launch successful attack in feasible time. We plot experimentally derived norm of shortest vector value in column $\|b_1\|'$, $\|b_1\|$ entries are for theoretically derived shortest vector value. It is calculated by the *i'* <*d* from experimental data and then plugging into the model for special case (section 6.1). We see that the experimental result is better than the theoretical one. We can reach upto *n = 27* within the time around 9.5 hours. We record the runtime to obtain pseudo-collision in both BKZ and BKZ-Progressive approach. Figure 4 shows the comparison graph.

C. *Pseudo*-Collision *in Typical case of* $\|b_i^*\|$

However, it might be interesting to see what experiment can achieve in perfect typical case condition.We found the following entries (in table 3) that satisfy the pseudo-collision constraint. To derive the value of q for corresponding larger

Table 3: Choice of parameters for finding *pseudo-collision* in typical case.

d	*m'*	*n'*	*P*	*k*	*q*
90	5	18	257	26	1.02538
100	1	25	79	28	1.02153
114	3	38	29	44	1.02210
120	3	40	29	50	1.02120

value of *k* we used linear extrapolation. In this case, it is hard to get a *pseudo-collision* as we cannot able to increase the value of $m' \leq m$ much (as its give shape like special case) and if we increase n we rarely find a short vector as volume of lattice jumps high. Again, in this particular situation we need to decrease modulus p to become successful.

The runtime for first two instances (d = 90 and 100) of BKZ reduction requires approximately 185 and 1170 seconds on average. On the other hand, BKZ-Progressive can reduce the instances for 92 and 575 seconds respectively on an average. So, the time improvement is about a factor of two. For other instances, as the block size and dimension both become increasingly high it is unexpected to get collision in feasible time.

REFERENCES

[1] V. Lyubashevsky, D. Micciancio, C. Peikert and A. Rosen. *SWIFFT: A Modest Proposal for FFT Hashing*. In Fast Software Encryption(FSE) 2008, LNCS, pp. 54-72. Springer-Verlag, 2008.

[2] A. K. Lenstra, H. W. Lenstra, Jr., and L. Lova´sz. *Factoring polynomials with rational coefficient s*. Math. Ann., 261(4): 515-534, 1982.

[3] C.-P. Schnorr. *A hierarchy of polynomial time lattice basis reduction algorithms*. Theoretical Computer Science, 53(2-3): 201-224, 1987.

[4] N. Gama and P. Q. Nguyen. *Predicting lattice reduction*. In EUROCRYPT, 2008, LNCS 4965, pp. 31-51. Springer, 2008.

[5] C.-P. Schnorr and M. Euchner. *Lattice basis reduction: improved practical algorithms and solving subset sum problems*. Math. Programming, 66:181-199, 1994.

[6] V. Shoup. NTL: A library for doing number theory. Available at http://www.shoup.net/ntl/.

[7] R. Kannan. *Improved algorithms for integer programming and related lattice problems*. In Proc. of 15th ACM Symp. on Theory of Computing (STOC), pp. 193-206. ACM, 1983.

[8] G. Hanrot, D. Stehle´. *Improved Analysis of Kannan's Shortest Lattice Vector Algorithm*. In CRYPTO 2007, LNCS 4622, pp. 170-186, 2007.

[9] N. Gama, P. Q. Nguyen, O. Regev. *Lattice enumeration using Extreme Pruning*. In EUROCRYPT, 2010, LNCS, Springer-Verlag, 2010.

[10] J. Buchmann and R. Lindner. *Secure Parameters for SWIFFT*. In INDOCRYPT'09, pp. 1-17, 2009. [11] R. Lindner and C. Peikert. *Better key sizes (and attacks) for LWE-based encryption*. In Proc. of the 11th international conference on Topics in cryptology, CT-RSA'11, pp. 319-339, Springer-Verlag, 2011.

[12] Y. Chen and P. Q. Nguyen. *BKZ 2.0: Better Lattice Security Estimates*. In ASIACRYPT'11, LNCS 7073, pp. 1-20.IACR, 2011.

[13] P. Q. Nguyen and V. Brigitte. *The LLL Algorithm: survey and*

application. Chapter-2 (Hermites Constant and Lattice Algorithms). ISBN 978-3-642-02294-4, 2010.

[14] C.-P. Schnorr. *Lattice reduction by random sampling and birthday methods*. In STACS'03, LNCS, vol. 2607, pp. 145,156. Springer, 2003.

[15] G. Hanrot, X. Pujol and D. Stehle´. *Analyzing Blockwise Lattice Algorithms using Dynamical Systems*. Accepted to CRYPTO 2011.

[16] C.-P. Schnorr and H. H. Ho¨rner. *Attacking the Chor-Rivest cryptosystem by improved lattice reduction*. In EURO-CRYPT95, LNCS 921, pp. 1-12,Springer-Verlag, 1995.

[17] D. Micciancio and O. Regev. *Post Quantum Cryptography*. Chapter Lattice based Cryptography. Springer-Verlag, 2009.

APPENDIX A

Table 4: Average enumeration time/block comparison for typical and special case (BKZ-Progressive Reduction, Prune = 1).

	Enumeration Time(ms)	
Block Size(k)	***Typical case***	***Special case***
34	0.086	0.033
38	0.16	0.06
42	0.41	0.27
46	1.36	0.50
50	5.26	2.35
54	23.33	7.62
58	145	48

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User-Authentication Approach for Data Security Between Smartphone and Cloud

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Abstract—**Cloud computing architecture provides a proper management to share distributed resources and services throughout the world via computer network. This architecture offers three main features, e.g. SaaS, PaaS and IaaS. And today's Smartphones are more compatible with this architecture, especially with IaaS because of its small storage capacity. Smartphones have become almost computer and these can be viewed as a miniature of personal computer. Since cloud computing share distributed data via network in the open environment so, there may occur security problems. To address this problem, this paper has proposed a new data security approach for Smartphone in cloud computing architecture, which ensures secured communication system and hiding information from others. Security level is maintained using the Global Positioning System (GPS) and network provider which ensures strong user-authentication to secure our cloud.**

Keywords: Cloud Computing; Smartphone; Data Security; RSA; El Gamal; DES; AES.

I. INTRODUCTION

In the recent time, cloud computing has become a most frequently talked and attractive phenomenon among the people all over the world. Though this is the newly introduced term in IT industries, but the fact inside it is more common and primitive technology of computer network. Today cloud computing has won people's heart and become appealing enough to collect billion of customers for itself. And the increasing number customers have become a major threat for cloud computing environment which results in security problem of customer's valuable data.

In cloud environment, resources are shared among all of the servers, but users don't know the fact that how their data are stored in the server. Because there is a lots of data of each user's stored in the server and user don't know which data is located in which database server and how [1]. That's why it is very easy for an intruder to access, misuse and damage the original form of data. For this it is also very much essential for the cloud to be secure through proper resource management [2]. Today, there is a variety of security models and algorithms are applied in the field of cloud computing. But, these become failed to meet all aspects security threats [3]. In E-commerce and online business, we need to involve high capacity security models in cloud computing. File encryption system based on AES is used in some thesis work [4]. DES based file encryption system is also worked out [5]. But their given models keep both the key and encrypted file in the same database server. Some thesis works have tried to recover this problem [6]. All of their approaches are suitable for some respective aspects but, don't cover all criteria to make a system secured.

Till now, we are talking about the data-security in the cloud computing architecture. But, what will be our secured approach if the user terminal is Smartphone [7]. Today's Smartphones have spurred a renaissance in mobile computing and can be viewed as a miniature of personal computer having all capabilities of it. As Smartphone is of limited storage and there is a fear of losing the phone, cloud can be a greater solution to this problem. But, there should be a well-defined approach for user-authentication of Smartphone and data-security in the cloud. This thesis work has proposed a new approach to data-security in Smartphone through stronger user-authentication.

II. METHOD

This thesis-work ensures data-security using two encryption algorithms:

- Public Key Encryption Algorithm
- Private Key Encryption Algorithm.

Here, in Fig. 1 public key encryption algorithm is used for prevent hacking of data in the communication line between the Smartphone and main server. That mean, it ensures secured channel for passing our data. And private key encryption algorithm is used for maintaining proper authorization of data and store encrypted data in the server so that the data will be useless if anyhow intruder get access to the database server. It should be mentioned that the decryption key of data will not be in the database server.

Stronger user-authentication is maintained using the decryption key of respective file and current location of the Smartphone using GPS (Global Positioning System).

Our proposed working model of cloud computing is depicted in Fig. 1. Here, we have made simulation to get appropriate algorithm using this cloud model and also deploy this model with more security with simulated data.

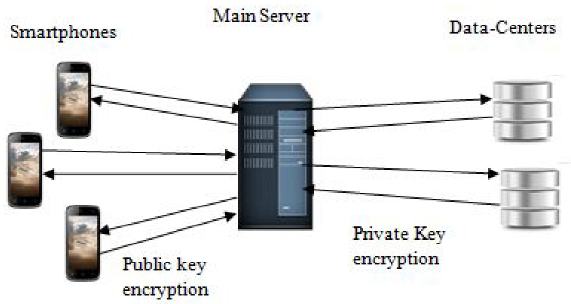

Figure 1. The Proposed Cloud Computing Model

In Fig. 1, we can see that the Smartphone user needs to contact with the main server to communicate with databases or datacenter. It takes files from the mobile users, stores these files in different databases, which are connected with it and retrieve files from the databases to specific users whenever needed. In our proposed model, user data is encrypted using public key cryptographic algorithm. That means user-data is encrypted in the Smartphone with the public key of the main server. Then server will decrypt the data with its private key. This public key cryptography ensures the security of data between the user and the main server. Now, come to the private key cryptographic algorithm which uses the same key for encryption and decryption. In our model this algorithm is used to store data in encrypted form in the database and send the decryption key to the Smartphone user's mail-box so that the data will be useless for the unauthorized persons. As a result, we can ensure a stronger user-authentication while downloading the data because to download data we need the decryption key of respective data. And decryption key is stored only in the authorized person's mail-box. So, even getting access to users cloud intruder can't get the meaningful data. There are many private key and public key cryptographic algorithms. Now, the problem is that which private key algorithm and public key algorithm we should use for our proposed model. In the next part we have made a simulation to get best fitted algorithm for our proposed approach to provide secured cloud environment.

III. SIMULATION TO GET SUITABLE ALGORITHMS

Smartphone, a little device with the promise of doing various computational tasks as like as personal computer is the terminal device for using the cloud computing environment. There is a major concern for computing devices that is power. Proper and efficient computing can ensure proper power management [8]. And as a small device with little battery power we should select such an algorithm which would not make our small computing device busy and consume a relatively small amount of time for doing the encryption.

A. Simulation Environment

- Processor Core 2 duo 2.8 GHz
- RAM 2GB
- Windows 7
- Android 4.0.4
- Java-script, PHP, MySQL Server

B. Simulation of Public Key Algorithms

There are many public key algorithms in cryptographic world. We select popular two from these which give better security and flexibility. These are:

- RSA public key algorithm
- El Gamal public key algorithm

RSA requires less energy for generation of keys, encryption and decryption of the data [9]. Vijayalakshmi shows in her paper work [10] that energy consumption in El Gamal is greater than in RSA algorithm for the same task.We select one smaller exponent (e) or encryption key in the RSA encryption. But El Gamal requires two exponentiations for encryption [11]. Fig. 2 shows RSA encryption requires less time than El Gamal but, Fig. 3 show the opposite. The file is encrypted on the mobile and decrypted on server. So, our goal is to minimize the encryption time whatever the total cryptographic time as depicted in Fig. 4. We get some experimental data (see Table 1) from our simulation on RSA and El Gamal public key algorithm.

Table 1. File Encryption & Decryption Time in RSA & El Gamal

File Size (KB)	RSA			El Gamal		
	Encry -ption Time (sec)	*Decry -ption Time (sec)*	*Total Time (sec)*	*Encry -ption Time (sec)*	*Decry -ption Time (sec)*	*Total Time (sec)*
1	0.07	0.19	0.26	0.13	0.11	0.24
2	0.13	0.38	0.51	0.27	0.21	0.48
3	0.20	0.58	0.78	0.40	0.33	0.73
4	0.27	0.78	1.05	0.52	0.44	0.96
5	0.34	0.97	1.31	0.66	0.56	1.22
6	0.40	1.15	1.55	0.79	0.67	1.46
7	0.47	1.35	1.82	0.93	0.78	1.71
8	0.54	1.54	2.08	1.05	0.88	1.93

C. Simulation of Private Key Algorithms

Among the various private key algorithm, DES and AES is main which apply more secure technique to make the system save. Both the encryption and decryption is occurred in server side. So, there is no need to separately consider the encryption and decryption time. Because, our prime concern is that Smartphone has to perform upload and download in a shorter cryptographic time. For this total time required to encrypt and decrypt the file is considered. Our simulated data in lab shows the following result (see Table 2) regarding DES and AES.

Table 2. Total Cryptographic Time (second) in DES and AES

File Size (KB)	DES Cryptographic Time (sec)	AES Cryptographic Time (sec)
1	0.03	0.05
2	0.06	0.09
3	0.08	0.14
4	0.11	0.18
5	0.13	0.23
6	0.16	0.28
7	0.19	0.33
8	0.21	0.38

In our proposed model, we are giving priority on the capability of the small device, Smartphone maintaining a standard security level. We compare execution time (encryption and decryption) of DES [12] with AES [13] depicted in Fig 5. We are not considering triple DES (3DES) here, because it has high execution time than DES and we already know that.

D. *Graphical Comparison*

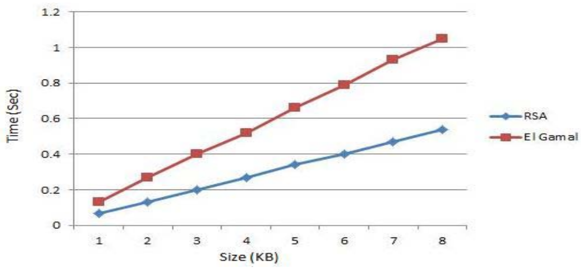

Figure 2. Encryption Time of RSA Versus El Gamal

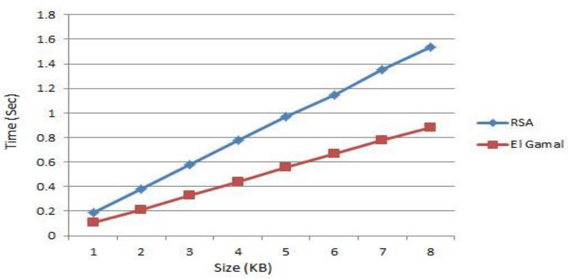

Figure 3. Decryption Time of RSA Versus El Gamal

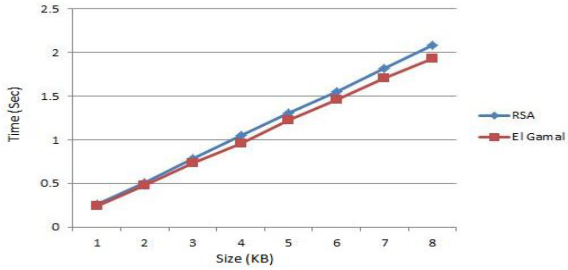

Figure 4. Total Cryptographic Time of RSA Versus El Gamal

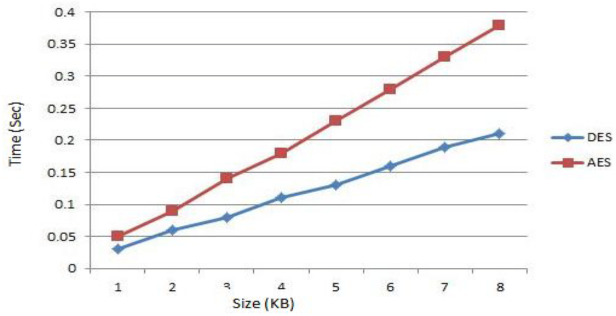

Figure 5. Total Cryptographic Time of DES Versus AES

IV. Deployment of the Proposed Model

After making a proper simulation for our proposed model, we think that RSA and DES cryptographic algorithm is best suited for Smartphone to access the cloud. We have select RSA for its less encryption time (Fig. 2) as encryption occurs in Smartphone and our aim is to decrease computational time in Smartphone.

A. *File uploading to cloud*

In Fig. 6 RSA public key algorithm encrypt user's file in the mobile device with the public key (k1-) of the main server. And main server will decrypt this file with its private key (k1+) and encrypt the file again with DES private key algorithm using symmetric key, k2- and send the decryption key (k2+) to the user's mail-box. So, only authenticated user can get useful file from the database.

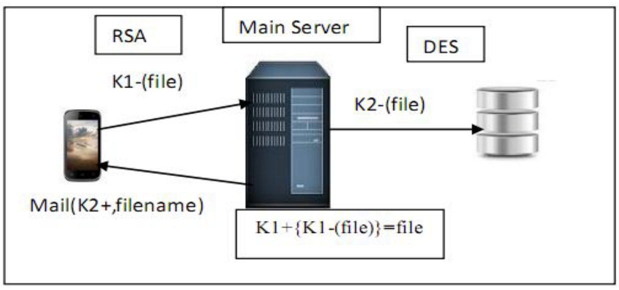

Figure 6. File Uploading to Cloud

B. *File Downloading from Cloud*

Fig. 7 describes the file downloading procedure. To download the files users have to login with their mail-id. One user may have many files in the cloud. So, users have to provide their corresponding file-name and DES decryption key (k2+) which was sent to their mail-box.

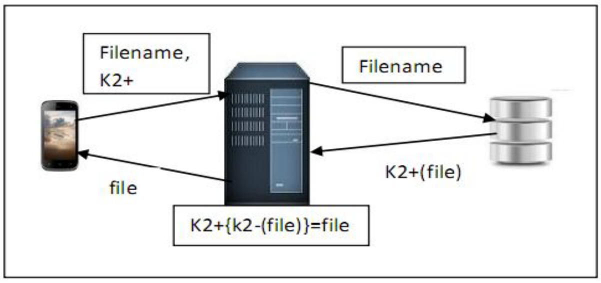

Figure 7. File Downloading from Cloud

C. *Ensuring Stronger User-Authentication to Secure User-data*

And finally we have tried to meet the cloud using mode of all users and provide them a more flexible way to get security at different level. Smartphone is the most personal device and users of it want Smartphone to be smarter reducing their own smartness or awareness. Users keep them always logged in the cloud in their Smartphone device to avoid several login. But, what will be happen to their cloud if the phone is lost or stolen intentionally to get access to personal data.

This is final step of deployment of our proposed model and this will provide greater flexibility to access a secure cloud from user's Smartphone offering different security mode. Here, we define three modes as the following:

1) Mode-1(Normal mode): In this mode we assume user is more smart and aware enough and logout each time of accessing cloud. So, he/she will follow the normal data-security model as described in file uploading and downloading section.

2) Mode-2(Secured Mode): This mode provides a high level security to the users. Here, a different database table (see Table 3) is maintained to ensure security.

Table 3. Database Table for Security Question

user_id (mail-id)	File-name	Security question	Answer (ans)
ab@yahoo.com	ac.txt otripto.txt ……… …….	MD5(My first day at school?)	MD5(anS)
……………	……… ………	………….	………..

The above table is maintained by the authorized user. There may be a situation that user has lost his phone and all accounts (cloud account and e-mail account) browser are logged in as he prefers to do it. Now, it is very easy for intruder to get access to the cloud account as the key containing mail account and cloud account both are open to him. Then what will happen to the user's valuable data. To answer this question we propose this secure mode. Users can set more anonymous question and answer in a database table (see Table 3) to restrict intruder to get data. However, intruder will not get the actual answer or can't guess any answer from question if he/she gets access to database because; MD5 Message-Digest algorithm [14] is applied here. So, this secured mode restrict intruder with a security question.

3) Mode-3(Advance mode): If the user is in the regular location (location will trace through GPS of Smartphone and network provider) then use normal mode else use secure mode. Each of us has some particular working locations from where we access our cloud. We define these locations as regular location. So, our proposed model provides a strong user-authentication and by this way which indirectly maintain our data security.

V. CONCLUSION

In this thesis paper, we have worked to get a suitable cloud computing security architecture for Smartphone. Smartphone is a mobile and small device. So, the general traditional security approaches will not fit to Smartphone-cloud architecture. Considering its computing power and battery power we proposed RSA and DES algorithm in this architecture. And in final implementation of our proposed model we have tried to make a strong user-authentication considering its mobility. Ensuring the stronger user-authentication we ensure the data-security between Smartphone and cloud.

REFERENCES

[1] M. A. Vouk, "Cloud Computing – Issues, Research and Implementations", *Journal of Computing and Information Technology - CIT 16*, vol. 4, pp. 235–246, 2008.

[2] Y. Hu, J. Wong, G. Iszlai, M. Litoiu, "Resource Provisioning for Cloud Computing", *IBM Canada Ltd.*, 2009.

[3] N. G. Mollet, "Cloud Computing Security", Thesis Paper, April 11, 2011.

[4] Y. Tang, P. P. C. Lee, J. C. S. Lui, and R. Perlman, "FADE: Secure Overlay Cloud Storage with File Assured Deletion", 2010.

[5] N. Jain and G. Kaur, "Implementing DES Algorithm in Cloud for Data Security", *VSRD-IJCSIT*, vol. 2 (4), pp. 316-321, 2012.

[6] T. S. Kar, M. A. P. Mahmud, "A Newer Secure Communication, File Encryption and User Identification based Cloud Security Architecture" *International Journal of Computer Applications (0975 – 8887)*, vol. 52– No.4, August 2012.

[7] (2013, Aug.) The Wichita education website. [Online]. Available: http://webs.wichita.edu/depttools/depttoolsmemberfiles/wines

/4A_Tandel_Venkitachalam_Cloud-Computing_Smartphones.pdf

[8] G. Dhiman and T. Š. Rosing, “System-Level Power Management Using Online Learning”, *IEEE Transactions On Computer-Aided Design Of Integrated Circuits And Systems*, vol. 28, no.5, May 2009.

[9] R. L. Rivest, A. Shamir, and L. Adleman, “A Method for Obtaining Digital Signatures and Public-Key Cryptosystems”, *Communications of the ACM*, vol. 21, Feb 1978.

[10] Kayalvizhi.R, Vijayalakshmi.M, Vaidehi.V, “Energy Analysis of RSA and ELGAMAL Algorithms for wireless Sensor Networks”, Proceedings of the 8th WSEAS International Conference on Applied Electronics, Wireless and Opticalal communication, 2012.

[11] T. El Gamal, “A public key cryptosystem and a signature scheme based discrete logarithms”, *HP labs*, 1985.

[12] R. G. Kammer, W. M. Daley, “Data Encryption Standard (DES)”, *Federal Information Processing Standards Publication*, FIPS PUB 46-3, 1999.

[13] J. Daemen, V. Rijmen, “Announcing the ADVANCED ENCRYPTION STANDARD (AES)”, *Federal Information Processing Standards Publication 197*, November 26, 2001.

[14] Ronald Rivest, “MD5 Message-Digest Algorithm”, *rfc 1321*, April 1992.

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A Binary Code Lock System

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***Abstract*— The proposed system is intended to protect electrical appliances with a simple lock system. As for passcode, a 4 bit binary number is used. To input the pass code, four two state switches are utilized. The pass code is stored in a memory. It can be changed after unlocking the lock system using valid pass code. We use D type flip-flops as memory. The comparator unit compares input codes with the stored code to generate the signal required to open the lock. The system is hardware implemented and has the versatility in application.**

Keywords: Code lock; Pass code;Comparator

I. INTRODUCTION

A locking device which is operated by electric current is called an electronic lock or more precisely electric lock. More often, electric locks are used for access control. A Code lock system is a kind of electronic lock system where a code is used for unlocking the lock. Only for the valid code the lock opens, otherwise it remains locked. A Binary code lock system is the one in which the code is a binary number instead of a decimal number. It makes the input process a lot simpler down to just opening or closing a series of switches. The valid code is stored in a memory element comparisons to which an unlock command is generated for a valid input code to open the lock.

There are several ways to implement binary coded lock system [1-5]. In [1], the proposed system is built by using only one 8-position DIP switch. But, this can also be implemented by two switch assemblies. In this approach, one switch acts to hold the correct code for unlocking the lock, while the other switch serves as a data entry point for the person trying to open the lock. A microcontroller based approach is implemented in [2] where an Atmel AT89C2051 (U1) microcontroller is used to build the system. Solenoid is controlled from a power MOSFET IRF540 (VT3). This additional transistor is useful as it translates the microcontroller unit logic levels to 0V and 12V, capable to drive the solenoid. A single integrated circuit based approached is implemented in [3], which requires a code of seven digits. In [4], the authors proposed a binary single-key-lock system for access control.

In this paper, we proposed a binary code lock system which is completely different than others. We used flip-flop memory and EX-OR gates as a comparator. A 4 bit binary number is used as a passcode. The stored passcode is compared with the input passcode to unlock the system. D flip-flop is used as memory to store the passcode and comparator unit compare input codes with the stored code to generate unlock command.

The rest of this paper is organized as follows. In section II, we introduce the design overview of our proposed system. In section III, we illustrate experimental results & discussions and finally we conclude this paper in section IV.

II. DESIGN OVERVIEW

In Fig.1, we show the block diagram of our proposed binary code lock system.

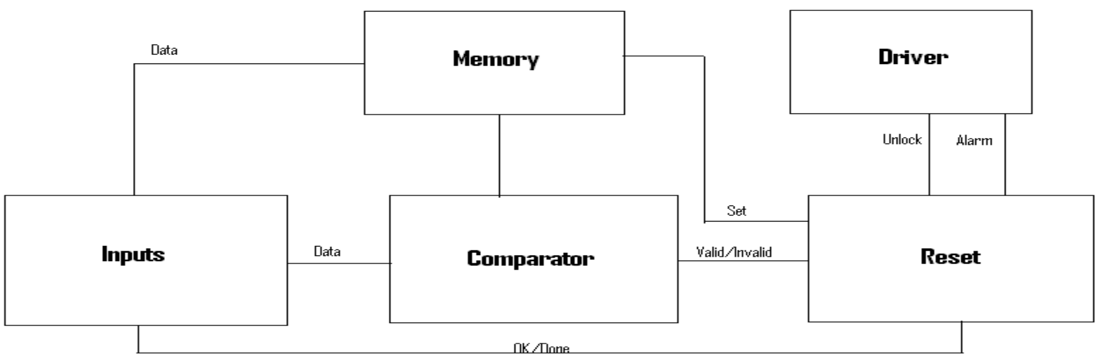

Figure 1. Block Diagram of the proposed binary code lock system.

A. Inputs

Four two state push button switches are used as data inputs which are shown in Fig. 2. They are used to input logic 0 or 1 to the memory as well as comparator. There are also three control inputs, which are Set, Done/Ok and Unlock.

- Set is used to change stored memory data.
- Done/OK is used to signal the completion of input.
- Unlock is used to open the lock.

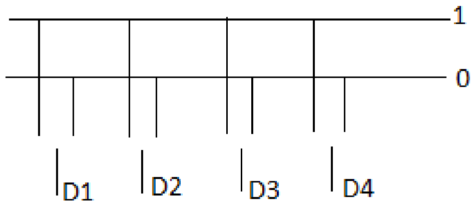

Figure 2. Inputs.

B. Memory

Four D flip-flops are used as memory to store four bit binary data shown in Fig. 3. Stored data can be changed using Set command which is actually a clock signal to the memory flip-flops. The outputs of the flip-flops are delivered to the comparator for decision making.

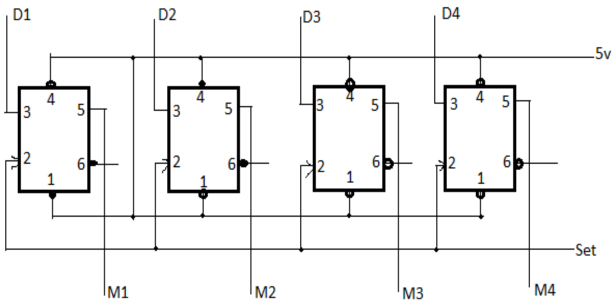

Figure 3. Memory circuit.

C. Reset

The reset unit is basically a D flip-flop shown in Fig. 4. Its data input is the inverted comparator output. OK/Done command stores the comparator output and the inverted output of the reset flip flop is used to generate Unlock and Set command which can be activated by corresponding control input.

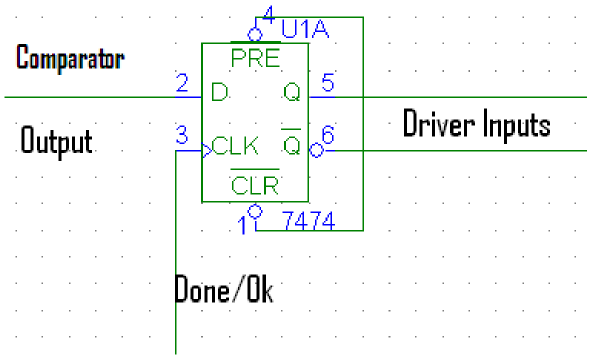

Figure 4. Reset circuit.

D. Comparator

The comparator unit is shown in Fig. 5. The input data and memory data are compared by four XOR gates. If an input-memory data pair matches, XOR output becomes 0 which is inverted and the inverted output of all four XOR gates are ANDed. Hence, if the input and memory code are identical, the comparator output goes high which is connected to the reset unit.

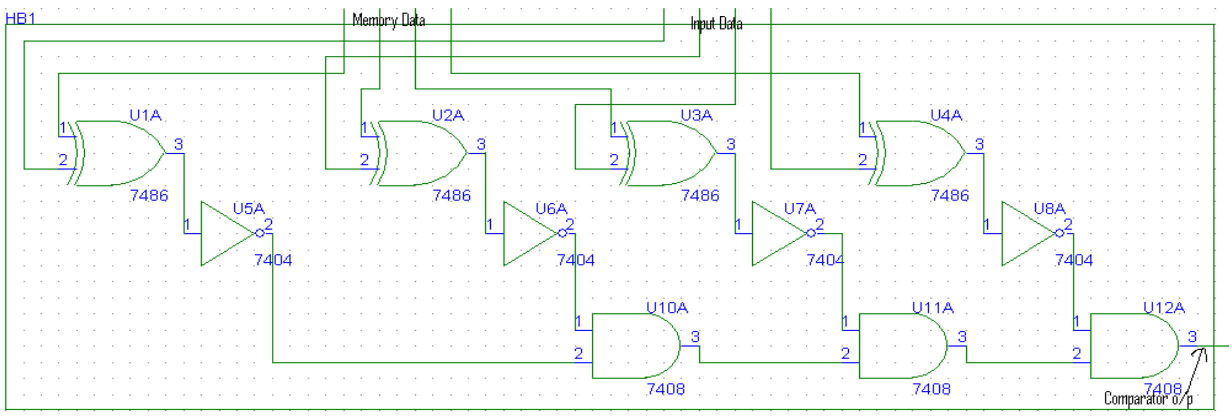

Figure 5. Comparator circuit.

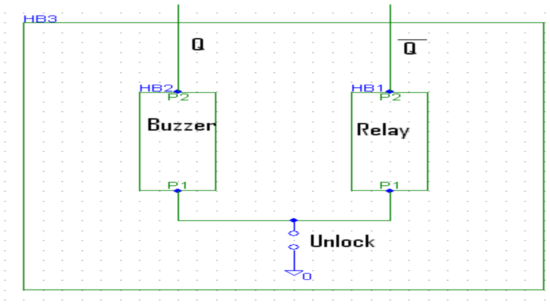

Figure 6. Driver circuit

E. Driver

The driver unit consists of a relay and a buzzer shown in Fig. 6. Both are activated by the unlock command. For a valid code, the inverted output of the reset flip-flop goes high which activates the relay. For an invalid code, the non-inverted output of the reset flip-flop goes high and sounds the buzzer.

F. Complete circuit diagram of the proposed system

In Fig. 7, we show the complete circuit diagram of the proposed system. The operation of the system can be described as follows:

- Initially the code is set to 1111 by default.
- To unlock for the first time, we input 1111 via data inputs.
- Then, we activate OK/Done control input.
- To change the code, we input the code that we want as the pass code via data inputs and activate Set input. Hence, the code has changed.
- To lock the system, we give any invalid code and activate OK/Done.
- Now, to unlock we must use the changed code otherwise activating Unlock input will trigger the alarm.
- With valid code activating Unlock input will turn on the relay and so the locked appliance.

III. Experimental Results and Discussions

We activate the lock using initial default code 1111. Then, we set the code to 1100. Afterward, we tried to open the lock using 1101 which sounds the buzzer. To turn the buzzer off, we use 1100 and this buzzer goes off and the relay is turned on. Then, we set the input to 0101 and press Set to change the code to 0101. Now, any code other than 0101 triggers the buzzer whereas only 0101 turn on the relay by pressing Unlock control input. However, the proposed system offers several advantages such as low complexity, low computational cost, easy installation, alarming unauthorized attempts to use electrical devices as well as can be used to control electricity usage thus saving electricity bill. Moreover, people with the zero technical knowledge can operate the system by following the instructions.

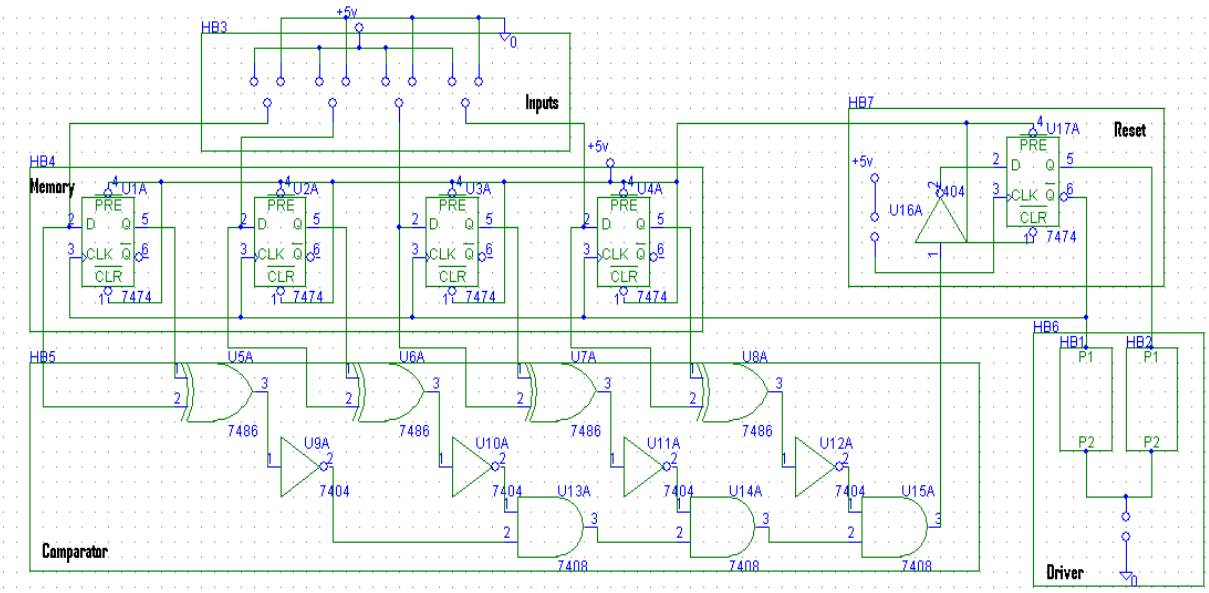

Figure 7. Complete circuit diagram of the proposed binary code lock system.

In addition, other than the advantages, it has some limitations such as less secure due to only 16 possible pass codes possible, adding additional bits for pass code exponentially increase the circuit complexity and difficult to troubleshoot. In our future work, we will consider these problems so that our proposed system can be available for commercial purposes.

IV. CONCLUSION

We successfully implemented the binary code lock system. In the proposed system, we utilize some basic components of digital electronics. The idea was to utilize something very simple and turn it into something of much greater use. With the proposed system, we can effectively secure our electrical appliances and control the use of valuable electrical energy in our home or office. It may be used to lock doors, cars and small safe's also with a little improvisation. Hence, it offers a wide range of applications and does not require any fancy component which decreases the cost significantly.

REFERENCES

[1] Simple Combination Lock, [Online]. Available: http://www.Learning- electronics. net/vol_6/ chpt_7/7.html

[2] iButton electronic lock, [Online]. Available: http://www.electronics-lab.com/projects/sensors/001/index. html

[3] Electronically Lock for Door by Keypad mini project, [Online]. Available: http://www.examsadda.com/2011/05/ electronically-lock-for-door-by-keybad.html

[4] C.K. CHANG AND T.M. JIANG, "A BINARY SINGLE-KEY-LOCK SYSTEM FOR ACCESS CONTROL," IEEE TRANSACTIONS ON COMPUTERS, VOL.38, NO.10, PP.1462-1466, 1989.

[5] Il-kyu Hwang, J. Baek,"Wireless access monitoring and control system based on digital door lock," IEEE Transactions on Consumer Electronics,vol. 53 , No.4,November 2007.

A Method for Pitch Detection of Speech Signal in Noisy Environment

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***Abstract*—An efficient pitch detection algorithm is proposed in this paper. The algorithm is based on time domain pitch detection algorithm. In our proposed method, instead of the original speech signal, we employ its center clipping signal for obtaining the autocorrelation function and this function is weighted by the reciprocal of the average magnitude difference function for pitch detection. The performance of the proposed pitch detection method is compared in terms of gross pitch error with the other related method. A comprehensive evaluation of the pitch estimation results on male and female voices in white noise show the superiority of the proposed method over the related method under low levels of signal to noise ratio (SNR).**

***Keywords*— Robust. Pitch, Center Clipping, White Noise.**

I. INTRODUCTION

The speech signal can be classified into two general categories, voiced and unvoiced speech. A voice sound is one in which the vocal cords of the speaker vibrate as the sound is made, and unvoiced sound is one where the vocal cords do not vibrate. The fundamental frequency of a voice sound, whose percept is called pitch, has great importance in many areas. Pitch detection is one of the oldest, yet unsolved topic among the researchers of speech and music [1,2]. Accurate pitch detection is essential to areas such as automatic speech recognition, speaker identification, low-bit rate coding, speech enhancement using harmonic model, speech synthesis, and to more recent topic of speaker emotion recognition etc. [3,4,5,6]. Recently many pitch estimation algorithms have been proposed, but accurate and efficient pitch estimation is still a challenging task.

There are three types of pitch detection algorithms (PDAs) in the literature: time domain [7, 8], frequency domain[9, 10], and time-frequency domain[11]. Due to the extreme importance of accurate pitch detection problem, the strengths of different PDAs have been explored [12,13], and several pitch reference databases have been developed to facilitate fair comparison of different PDAs on a common platform [14]. Among the reported method, the time domain method i.e., autocorrelation function (ACF) [7] based approaches are very popular for their simplicity, low computational complexity, and good performance in the presence of noise. The ACF is, however, the inverse Fourier transform of the power spectrum of the signal. Thus if there is a distinct formant structure in the signal, it is maintained in the ACF. Spurious peaks are also sometimes introduced in the spectrum under noisy or even under noiseless conditions. This sometimes makes true peak selection a difficult task. This motivates researches to propose numerous modifications on the ACF method. One significant improvements are proposed in Sondhi [15] used center clipping ACF. On the other hand, a well known method, average magnitude difference function (AMDF) has the advantage of low computation and high precision and the calculation cost needed less than that of ACF [16]. But when the magnitude or the pitch period of speech signal changes rapidly, AMDF method will decreased apparently in pitch estimation accuracy. Correlation based processing method is known to be comparatively robust against noise. In this paper, we proposed a correlation based pitch detection method, where the centre clipping signal is used for ACF and this ACF is weighted by the inverse of an AMDF. The proposed method utilizes the feature that in a noisy environment, the noise components included in the ACF and AMDF behave independently. By such type of uncorrelated properties, the peak of the ACF is emphasized when the ACF is combined with inversed AMDF [17].

The paper is organized as follows: Section II describes some basic pitch detection algorithms that include time domain processing. Section III presents the proposed pitch detection algorithm, and Section IV gives experimental results. Finally, the paper is concluded in Section V.

II. PITCH DETECTION ALGORITHMS

Pitch or fundamental frequency is the lowest frequency component of a signal that excites to a system i.e., vocal system. The pitch period, which is the inverse of fundamental frequency, is the smallest repeating unit of a signal. One such period describes the periodic signal (i.e., voiced part of speech) completely. The fact that variations in voiced signal are so evident suggests that the time domain method should be capable in detecting pitch period of a voiced signal. Most of the time domain pitch period estimation methods use ACF.

Let $x(n)$ and $v(n)$ denote speech signal and uncorrelated white Gaussian noise with zero mean and variance σ^2_v, respectively. Therefore, the noisy signal $y(n)$ is then given by

$$y(n) = x(n) + v(n) \tag{1}$$

Based on the assumption that speech and noise are uncorrelated, the ACF $R_{yy}(k)$ of $y(n)$ can be expressed as

$$R_{yy}(k) = \begin{cases} R_{xx}(k) + \sigma_v^2 & \text{for } k = 0, \\ R_{xx}(k) & \text{for } k \neq 0, \end{cases} \tag{2}$$

where $R_{xx}(k)$ is the ACF of the clean speech signal $x(n)$ estimated as

$$R_{xx}(k) = \frac{1}{N} \sum_{n=0}^{N-1} x(n)x(n+k) \tag{3}$$

here N is the total number of samples in a window of the speech under analysis and k is the lag index. The choice of window length N for calculating $R_{xx}(k)$ has conflicting requirements:

- N should be as small as possible to show time variation;
- N should be large enough to cover at least 2 periods so that periodicity can be captured by $R_{xx}(k)$.

In Eq. (3), $R_{xx}(k)$ essentially exhibits peaks at the periodicity (T) of $x(n)$ (i.e., at $k=lT$, where l is an integer). The basic idea behind the ACF based methods is to use the location of the second largest peak (at $k=T$) relative to the largest peak (at $k=0$) to obtain an estimate of the pitch period (Fig. 1). The main advantage of ACF method is its noise immunity. However, it effects the formant structure which result in the loss of a clear peak in $R_{xx}(k)$ at the true pitch period. The performance of the conventional ACF method is significantly degraded at low SNR (Fig. 2).

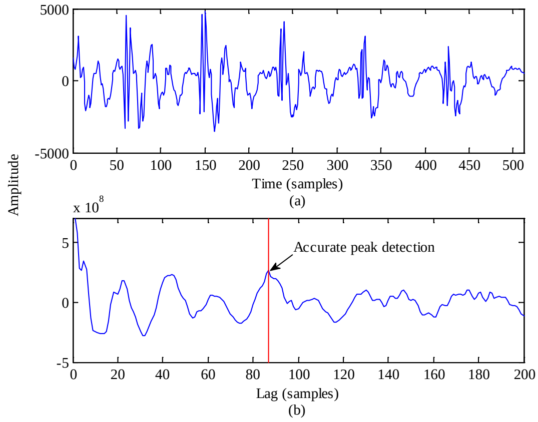

Figure 1: (a) Clean speech signal of a male speaker, (b) Autocorrelation function of signal in (a). The vertical line indicates the correct pitch value.

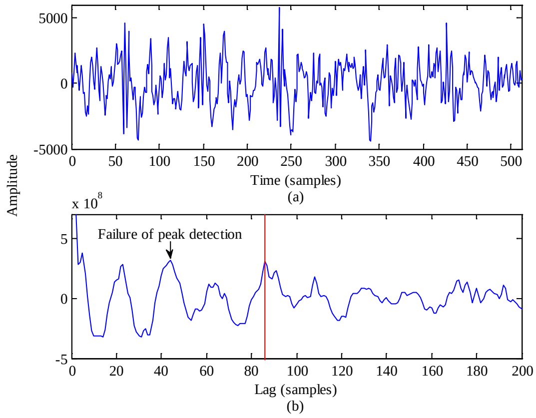

Figure 2: (a) Noisy speech signal of male speaker (which is the same frame as Fig. 1(a)) at an signal to noise ratio of -5dB, (b) Autocorrelation function of signal in (a). The vertical line indicates the correct pitch value.

The average magnitude difference function (AMDF) is another type of autocorrelation analysis. Instead of correlating the input signal at various delays, a difference signal is formed between the delayed signal and original, and at each delay value the absolute magnitude is taken. The AMDF is describe by

$$\xi_{xx}(k) = \frac{1}{N} \sum_{n=0}^{N-1} \left| x(n) - x(n+k) \right| \tag{4}$$

where $x(n+k)$ are the samples time shifted on k samples. The difference function is expected to have a strong local minimum if the lag k is equal to or very close to the fundamental frequency. AMDF has advantage in relatively low computational cost and simple implementation. Unlike the autocorrelation function, the AMDF calculations require no multiplications. This is a desirable property for real time applications. For each value of delay, computation is made over an integrating window of N samples. The fundamental frequency period is identified as the value of the lag at which the minimum AMDF occurs (Fig. 3).

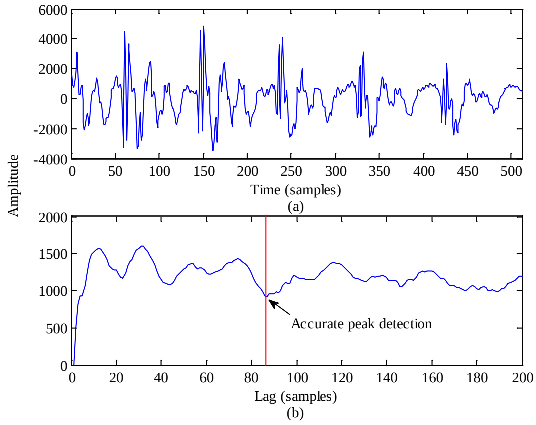

Figure 3: (a) Clean speech signal of a male speaker (which is the same frame as Fig. 1(a)), (b) Average magnitude difference function of signal in (a). The vertical line indicates the correct pitch value.

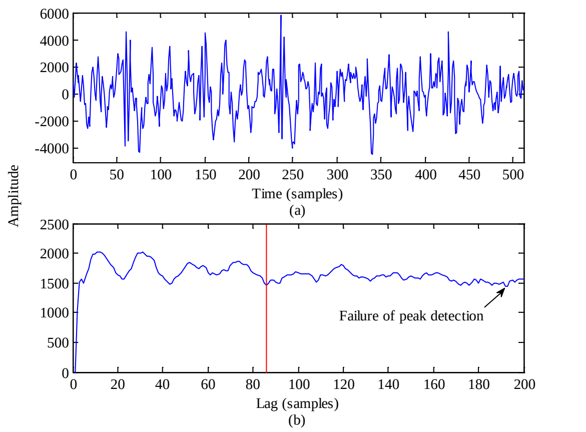

Figure 4: (a) Noisy speech signal of a male speaker (which is the same frame as Fig. 1(a)), (b) Average magnitude difference function of signal in (a). The vertical line indicates the correct pitch value.

This algorithm has many advantages, but the probability of double misjudge and half misjudge is very high when noise is added (Fig. 4).

III. PROPOSED METHOD

The ACF weighted by the inverse of an AMDF is used for fundamental frequency extraction [17] and is defined as

$$\varphi_{xx}(k) = \frac{R_{xx}(k)}{\xi_{xx}(k) + \delta} \tag{5}$$

where $R_{xx}(k)$ and $\xi_{xx}(k)$ denotes the ACF and AMDF of signal $x(n)$ respectively, δ is a small positive constant. It is expected to give maximum peak at $k = nT$ (ACF) & deep notches at $k = nT$ (AMDF), and therefore the true fundamental frequency peak in $\varphi_{xx}(k)$ is emphasized (Fig. 5). But the main limitation of this method is that, it is very sensitive to the half or double pitch error in noisy case as shown in Fig. 6.

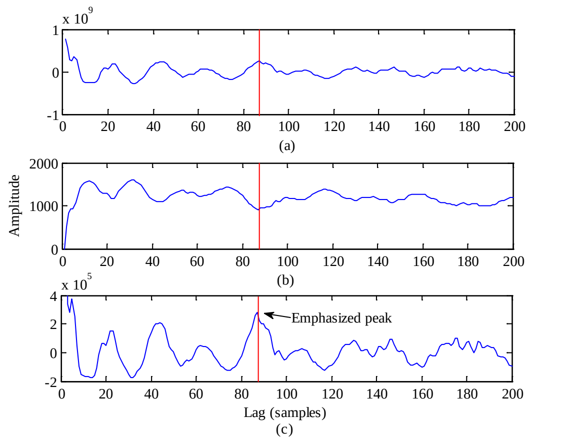

Figure 5: Pitch peak detection in clean speech signal (which is the same frame as Fig. 1(a)) using (a) Autocorrelation function method, (b) Average magnitude difference function method, (c) Weighted autocorrelation function method. The vertical line indicates the correct pitch value.

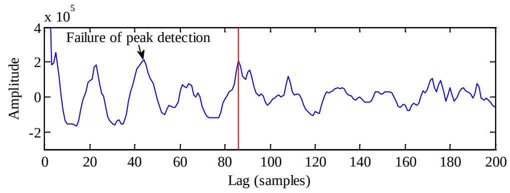

Figure 6: Pitch peak detection in noisy speech signal (which is the same frame as Fig. 1(a)) using Weighted autocorrelation function method. The vertical line indicates the correct pitch value.

For fundamental frequency detection, speech signal is usually pre-processed to make the periodicity more prominent and to suppress other distracting features. Such techniques are often called spectrum flattening. Center clipping is the most popular spectrum flattening technique [15] and we used this technique in our proposed method. Center clipping technique can be expressed as

$$x'(n) = C_L\{x(n)\} = \begin{cases} (x(n) - C_L), & x(n) \geq C_L \\ 0, & |x(n)| < C_L \\ (x(n) + C_L), & x(n) \leq -C_L \end{cases} \tag{6}$$

where $x'(n)$ is the center clipping signal of speech signal $x(n)$ and C_L is the clipping level. A choice of C_L should be fulfill the following criterion:

- should be high enough to eliminate all distracting peaks, but
- cannot be too high so as not to lose desirable peaks.

In our proposed method, instead of the speech signal $x(n)$, we employ its center clipping signal $x'(n)$ for obtaining the ACF and using this ACF weighted by $1/\xi_{xx}(k)$. It is expected that the true peak is more emphasized (Fig. 7), and as a result the errors of fundamental frequency extraction are decreased. The correlation based proposed method is given by

$$\varphi_{xx\_cl}(k) = \frac{R_{xx\_cl}(k)}{\xi_{xx}(k) + \delta} \tag{7}$$

where $R_{xx\text{-}cl}(k)$ is the ACF of signal $x'(n)$.

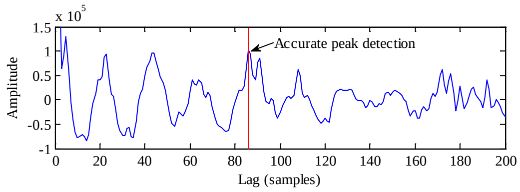

Figure 7: Pitch peak detection in noisy speech signal (which is the same frame as Fig. 1(a)) using Proposed method. The vertical line indicates the correct pitch value.

IV. Experimental Results

To assess the proposed method, natural speeches spoken by two Japanese male and two Japanese female speakers are examined. Speech materials are 11 sec-long sentences spoken by every speaker sampled at 10 kHz rate, which are taken from NTT database [18]. The reference file of the fundamental frequency of speech is constructed by computing the fundamental frequency every 10 ms using a semi-automatic technique based on visual inspection. The simulations were performed after adding additive noise to these speech signals. For the performance evaluation of the proposed method, criteria considered in our experimental work is gross pitch error (GPE). The evaluation of accuracy of the extracted fundamental frequency is carried out by using

$$e(l) = F_t(l) - F_e(l) \tag{8}$$

where $F_t(l)$ is the true fundamental frequency, $F_e(l)$ is the extracted fundamental frequency by each method, and $e(l)$ is the extraction error for the l-th frame. If $|e(l)| > 20\%$, we recognized the error as a gross pitch error (GPE) [19,20]. Otherwise we recognize the error as a fine pitch error (FPE). The possible sources of the GPE are pitch doubling, halving and inadequate suppression of formants to affect the estimation. The percentage of GPE, which is computed from the ratio of the number of frames (F_{GPE}) yielding GPE to the total number of voiced frames (F_v), namely,

$$GPE(\%) = \frac{F_{GPE}}{F_v} \times 100 \tag{9}$$

As metrics, the GPE (%) provide a good description of the performance of a pitch estimation method. The experimental conditions are tabulated in Table I.

We attempt to extract the pitch information of clean and noisy speech signals. All the candidate algorithms are applied in additive white Gaussian noise. The noises are taken from the Japanese Electronic Industry Development Association (JEIDA) Japanese Common Speech Corporation. The performance of the proposed method is compared with a well-known weighted autocorrelation method, WAC [17]. For the implementation of the WAC, the parameter k in [17] is set to 1 and for proposed method, the parameter C_L is set to 10% of the maximum magnitude of signal. As the pitch range is known to be 50-500 Hz for most male and female speakers and our sampling frequency is 10 KHz, the setting of lag number (i.e., 200) is commonly used for the WAC and the proposed method.

Table I. Experimental Parameter Specification

Sampling frequency	10 kHz
Band limitation	3.4 kHz
Window function	Rectangular
Window size	51.2 ms
Frame shift	10 ms
Number of FFT points	1024
SNRs (dB)	∞, 20, 15, 10, 5, 0, -5

Pitch estimation error in percentage, which is the average of GPEs for male and female speakers, are shown in Fig. 8(a) and 8(b), respectively. These figure implies that the proposed method gives far better results for both male and female cases in different types of SNR conditions. These experimental results show that the proposed method is superior to the WAC method in almost all cases. Particularly, at low SNR (0 dB, -5 dB), the proposed method performs more robustly compared with the WAC method.

V. Conclusion

In this paper, an efficient fundamental frequency estimation using correlation based method was introduced which leads to robustness against additive noise. Simulation results indicate that the proposed method provides better performance in terms of GPE (in percentage) compared with the existing method such as WAC. These results suggest that the proposed method can be a suitable candidate for extracting pitch information in white noise conditions with very low levels of SNR as compared with other related method.

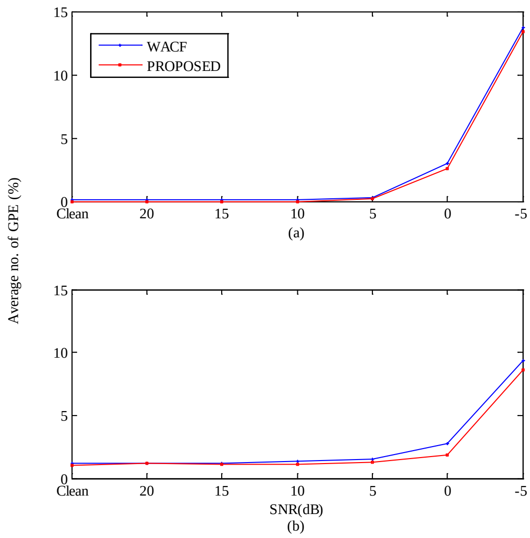

Figure 8: Percentage of average gross pitch error (GPE) for different speakers under various signal to noise ratio conditions; (a) Male speakers, (b) Female speakers.

REFERENCES

[1] W. Hess, "Pitch Determination of Speech Signals," Springer-Verlag, 1983.

[2] L. R. Rabiner, and R. W. Schafer, "Theory and Applications of Digital Speech Processing," 1st ed., Prentice Hall, 2010.

[3] H. Beigi, "Fundamental of Speaker Recognition," Springer, 2011.

[4] A. E. Rosenberg, and M. R. Sambur, "New techniques for automatic speaker verification," *IEEE Trans. Acoustics, Speech, and Signal Processing*, vol. ASSP-23, no. 2, pp. 169-176, 1975.

[5] M. Tamura, T. Masuko, K. Takuda, and T. Kobayashi, "Adaptation of pitch and spectrum for HMM-based speech synthesis using MLLR", In Proceedings of IEEE International Conference on Acoustic, Speech and Signal Processing (ICASSP'01), pp. 805-808, 2001.

[6] A. A. Razak, M. I. Z. Abidin, and R. Komiya, "Emotion pitch variation analysis in Malay and English voice samples", In Proceeding of the 9th Asia-Pacific Conference on Communications (APCC'03), vol. 1, pp. 108-112, 2003.

[7] L. R. Rabiner, "On the use of autocorrelation analysis for pitch detection," *IEEE Trans. Acoustics, Speech, and Signal Processing*, vol. ASSP-25, no. 1, pp. 24-33, 1977.

[8] M. A. F. M. R. Hasan, and T. Shimamura, "An efficient pitch estimation method using windowless and normalized autocorrelation functions in noisy environment," *International Journal of Circuits, Systems and Signal Processing*, Issue 3, vol. 6, pp. 197-204, 2012

[9] A. M. Noll, "Cepstrum pitch determination," *Journal of Acoust. Soc. Am.*, vol. 41, no. 2, pp. 293-309, 1967.

[10] S. Ahmadi and A. S. Spanias, "Cepstrum based pitch detection using a new statistical V/UV classification algorithm," *IEEE Trans. Speech and Audio Processing*, vol. 7, no. 3, pp. 333-338, 1999.

[11] M. A. F. M. R. Hasan, M. S. Rahman, and T. Shimamura, "Windowless autocorrelation based Cepstrum method for pitch extraction of noisy speech," *Journal of Signal Processing*, vol. 16, no. 3, pp. 231-239, 2012.

[12] L. R. Rabiner, M. J. Cheng, A. M. Rosenberg, and C. A. McGonegal, "A comparative performance study of several pitch detection algorithms," *IEEE Trans. Acoustics, Speech, and Signal Processing*, vol. ASSP-24, no. 5, pp. 399-417, 1976.

[13] P. Veprek, M. S. Scordilis, "Analysis, enhancement and evaluation of five pitch determination techniques," *Speech Communication*, vol. 37, pp. 249-270, 2002.

[14] F. Plante, G. Meyer, and W. A. Ainsworth, "A pitch extraction reference database", In Proceeding of EUROSPEECH, pp. 837-840, 1995.

[15] M. M. Sondhi, "New methods of pitch extraction," *IEEE Trans. Audio Electroacoust.*, vol. AU-16, pp. 262-266, 1968.

[16] M. J. Ross, H. L. Schafer, A. Cohen, R. F. B, and H. Manley, "Average magnitude difference function pitch extraction," *IEEE Trans. Acoustics, Speech, and Signal Processing*, vol. ASSP-22, no. 5, pp. 353-362, 1974.

[17] T. Shimamura, and H. Kobayashi, "Weighted autocorrelation for pitch extraction of noisy speech," *IEEE Trans. on Speech and Audio Processing*, vol. 9, no. 7, pp. 727-730, 2001.

[18] NTT, "Multilingual Speech Database for Telephometry," NTT Advance Technology Corp., Japan, 1994.

[19] A. Cheveigne, and H. Kawahara, "YIN. a fundamental frequency estimation for speech and music," *Journal of Acoust. Soc. Am.*, vol. 111, no. 4, pp. 1917-1930, 2002.

[20] M. K. Hasan, S. Hussain, M. T. Hossain, and M. N. Nazrul, "Signal reshaping using dominant harmonic for pitch estimation of noisy speech," *Signal Processing*, vol. 86, pp. 1010-1018, 2006.

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MOBILE ROBOT CONTROLLED BY VOICE COMMAND BASED ON GCM TECHNOLOGY USING ANDROID DEVICE

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***Abstract*—Mobile robot controlled by voice command means a robot that can be operated by our human voice. We have used voice recognition software to identify the appropriate command for robot to responds and another android device at the command sending end to fulfill our need. In a form of a short message a robot is capable of receiving the command and performs the predefined task according to the command. A micro-controller is added in the system to read the command given from the android device and perform the task accordingly such as move forward, backward, left, right and stop. We are using cloud messaging technique using GCM technology. Because of using the GCM technology our robot can be controlled even without being in front of the robot. We can give command sitting far away from the robot using internet. PHP server is used here for the 3rd party server.**

Keywords—GCM; Android; Micro controller; PHP; Robot.

I. INTRODUCTION

A mobile robot is an automatic machine that is capable of moving in any given environment. It can move around in their environment and are not fixed to one physical location. Robotic arms are relatively easy to build and program because they only operate within a confined area. Things get a bit trickier when you send a robot out into the world. As the technology advanced, scientists tried to build a robot that can actually move around its environment. To build a mobile robot, a set of instructions is needed which is passed to the micro-controller. Mobile robot is a combination of hardware and software. Different types of programming are needed.

To take this technology to a new level we are using another technology GCM (Google cloud messaging) on android device. Google Cloud Messaging (GCM) is a service that helps developers to send data from servers to their Android applications on Android devices, or from servers to their Chrome apps and extensions. The Android service was first unveiled on June 27, 2012, at Google I/O 2012 held at the Moscone Center in San Francisco. [1] This gives us an opportunity to operate the mobile from anywhere we want.

The objective of our project is to implement a cost effective and efficient movable and easily controllable mobile robot that can be controlled via android device from GCM technology. Google just announced that Google Cloud Messaging - the push-notification system that debuted last year, is now a part of its Google Play Services. We want to explore the robotics science with the latest technology like GCM.

These types of robot will be very beneficial to the handicap or physically disabled person. As it is dependent on the internet as well we can give command sitting anywhere away from the robot. Those who can't move they can easily move around with the help of this robot simply pressing a button. In different commercial and scientific field and even in the field of war, mini mobile robot can be operated by using internet without even being present in the war field. There are many helpful applications available of our robot.

II. FAMILARIZATION WITH GCM

In this section we have described GCM in short in sub-section A along with its characteristics in sub-section B and lastly we have described the role of the 3rd party on the sub-section C.

A. GCM

GCM stands for Google Cloud Messaging. It is a free service that allows developers to send data from third party servers to their applications running on Android devices. It handles queuing of messages and delivery to the target application running on the target device. Intended use is not to send a huge amount of data to the client device.

B. Characterteristics

- Allows 3rd-party application servers to send messages to their Android applications.
- An Android application on an Android device doesn't need to be running to receive messages. The system will wake up the Android application via Intent broadcast when

the message arrives, as long as the application is set up with the proper broadcast receiver and permissions.

- It does not provide any built-in user interface or other handling for message data.
- It requires devices running Android 2.2 or higher that also have the Google Play Store application installed, or an emulator running Android 2.2 with Google APIs.
- It uses an existing connection for Google services.
- Faster, Easier GCM Setup Streamlined registration makes it simple and fast to add GCM support to the Android app. Upstream messaging over XMPP. [2]

C. Role of 3rd party application server:

Before writing client Android applications that use the GCM feature, we must have an application server that meets the following criteria:

- Able to communicate with the client.
- Able to fire off HTTPS requests to the GCM server.
- Able to handle requests and resend them as needed, using exponential back-off.

III. RELATED WORK

There were different implementations in past for controlling a mobile robot. Most of them used PC based architecture and desktop application to control the system. Some implementation adopted GSM technology. It gives a lot of advantage than using costly RF transmitter to increase operating range. But you have to be dependent on third party GSM provider for this purpose.

In 2004, Yu, Chen, Cheng and H.H. proposed "Web based control system design and analysis" which was based on Ch (Which is a C/C++ interpreter) [3]. But it could not provide the vast amount of flexibility to its users. Shon presented “Protocol Implementations for Web Based Control Systems” in 2005 [4]. But he did not provide any mobility to that system. Pelko, Zagar, Zambon and Green presented “Canone - a highly interactive web-based control system interface” [5] in 2007 which was a critical web-based control system interface which lacked the mobility but provided a highly interactive system. Currently, much of the research is dealing with the wireless communication of robots. Cardozo, Guimaraes, Rocha, Souza, Paolieri, and Pinho, proposed “A Platform for Networked Robotics,” in 2010 [6] which focuses on the successful navigation and interaction of mobile robots rather than the flexibility. Now a day, emphasis are given more on these wireless control systems [7-8] than many other probable solutions. A Bluetooth technology was integrated with an android phone by Delden and Whigham in 2012 [9] to receive the command and control a robot. But the major limitation of that project was the limited range of Bluetooth Radio.

This paper focuses on controlling the movement of a mobile robot by android device which uses the GCM technology. Cloud Robot was always been a hot cake of this century. We are using GCM technology to pass the command to the microcontroller and thus control the robot. This system can be a good alternative to those traditional systems. This system is also cost effective than those using RF transmitter. This system is very flexible and easy to build. Its operating range is substantially as vast as Cellular Network. Moreover, it gives us the opportunity to use the various options of the majestic android platform.

IV. IMPLEMENTING ANDROID APPLICATION

In this section we have described the application used on the user’s device on sub-section A along with application of the robot’s device and PHP server work sequentially on sub-section B and C.

A. Application on the user’s device

1. *Voice control:* Here we have created a method called **dataentry.** It has the baseurl along with a value on the basis of the given command.

```
private void dataentry(String msg) {
String baseurl =
"http://fahadb.in/gcm_server_php/send_message.php?regId=APA91bH81-gSmRRm9pqowjOmrgi9-LCv85YmRzeTVehgMdccCeYiRqWlcWNzltowxJ4Muik1LOLRfDtapabRXDJZYIrVrjiJ4sF8 R0-YuDNioqR8TJLy_cnn-7VqGJSTGS7manjmVpmuJEalhgdNWq5b3Spya2HewTCeGrBnZ34Go-9kPAVGFtg&msg="+msg;
new SaveMsg(this).execute(baseurl);
}
    if(results.contains("right")){
        command = "1";
      dataentry(command);
   } else if(results.contains("left")){
        command = "2";
      dataentry(command);
  } else if(results.contains("forward")){
        command = "3";
      dataentry(command);
 } else if(results.contains("backward")){
        command = "4";
```

```
dataentry(command);
} else if(results.contains("stop")){
command = "5";
dataentry(command); }
```

Figure 1. voice command

B. *Application on the robot device:*

1) Installing helper libraries

Before start writing android code we need to install the helper libraries and make required changes to the emulator.

a. We had to install Google Cloud Messaging for Android Library under Extras section

b. After installing the library it has created gcm.jar file in our Andoird_SDK_Folder\extras\google\gcm\gcm-client\dist. Later we had added this .jar file to our android project.

2) Creating Android Project

Android development [10-11] completely revolves around “activities.” An android program usually contains multiple activities and these are found to be related to each other in most of the cases. It usually creates a window to interact with the user.

a) We had to create a new android project selecting minimum SDK version API 8 to support wider range of devices.

b) We need to add the following permission into our AndroidManifest.xml file to make our project support gcm.

INTERNET – To make your app use internet services ACCESS_NETWORK_STATE – To access network state (used to detect internet status) GET_ACCOUNTS – Required as GCM needs google account WAKE_LOCK – Needed if your app need to wake your device when it sleeps VIBRATE – Needed if your support vibration when receiving notification Also add some broadcast receivers as mentioned below.

c) We had created a class AlertDialogManager.java which is used to show alert dialog in the application.

d) Another class ConnectionDetector.java is used to detect internet connection status.

e) CommonUtilities.java class contains the GCM configuration and our server registration url.

f) We had created a new class called ServerUtilities.java with following content.

g) GCMIntentService.java class handles all GCM related services.

h) RegisterActivity.java class will be used to handler user registration.

C. *PHP server:*

3) Creating MySQL Database

a. First we had created a database called gcm.

b. After creating the database, we executed the following query in SQL tab to create **gcm_users** table.

4) Creating and running PHP project

a. When we are making request to GCM server using PHP we used curl to make post request. Before creating php project we had to enable curl module in our php extensions.

b. Then we had created a file called *config.php* This fill holds the database configuration and google api key.

define("GOOGLE_API_KEY", "AIzaSyCmlB5YQW_ipcv2CyDAaSvHL3 UAIoAQQ3s"); // this is our Google API Key

c. Then we create another file called *db_connect.php* This file handles database connections, mainly opens and closes connection.

d. *db_functions.php* file contains function to perform database CRUD operations. But we wrote the function for creating user only.

e. Then we create another file named *GCM.php* This file is used to **send push notification requests** to GCM server.

f. *register.php* file receives requests from android device and stores the user in the database.

g. *send_message.php* file used to send pushnotification to android device by making a request to GCM server.

Figure 2. PHP log in page

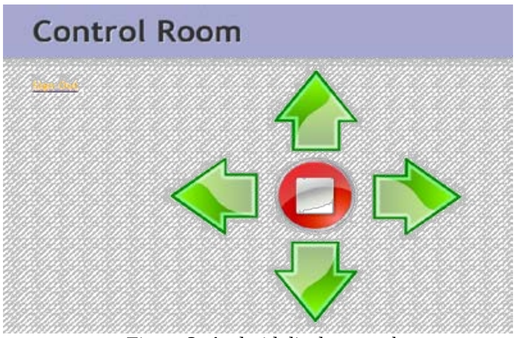

Figure 3. Android display on robot

V. Hardware Implementation

We have given a short description of our robot in sub-section A. Component we used are given in sub-section B. Circuit description and operation is shown in sub-section C and D. PCB layout along with the actual body of our robot is shown in sub-section E.

A. *Introduction:*

The hardware is basically a DC motor control board. The objective is to sense the color from android display and control the direction of rotation accordingly.

B. *Component needed:*

- LDR
- Potentiometer
- LM358
- ATMega16controller
- L298 Motor driver
- 7805 Voltage Regulator
- 1N4007 Diode

C. *Circuit description:*

1) LDR Input Section

In LDR input section, there are 4 LDRs in contact with the android display. Each LDR is connected with a POT in series and the other end of POT is connected to one of the negative inputs of LM358. Each LM358 has two pairs of input. Two LM358 have been used, with two LDRs connected to each.

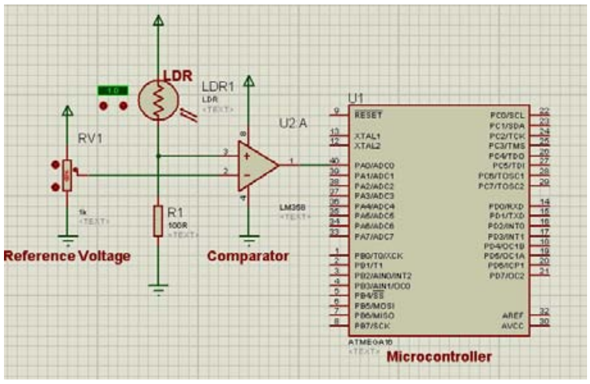

Figure 4. LDR Input Section

2) Micro-controller Section

An output from LM358 is connected to the inputs of ATMega16. The microcontroller is the processing and main controlling unit. Output of ATMega16 is connected to the motor driver L298.

3) Motor Drive Section

L298 is the motor driver. It takes input from the microcontroller and controls the two DC motors at the wheels.

The motor driver IC needs two voltage supply. One is Vs which is required for powering the motors and other is Vcc which is required for logical operations. One single supply of 12 volts is used in the circuit. 7805 voltage regulator steps down the 12 V to 5V for the Vcc supply.

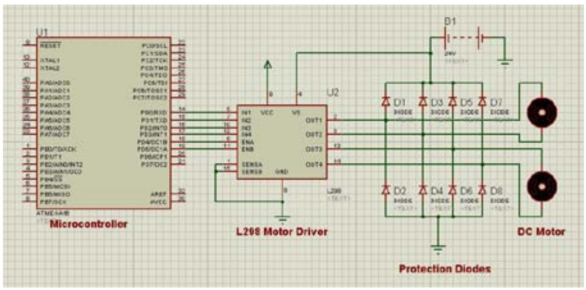

Figure 5. Motor Driver Section

D. *Circuit Operation:*

In the android display, there are four circular spots. These spots alternate their colors as blue or red. The resistance of LDR is changed with the change in color.

This change in resistance is used to divide a voltage of 5 V using a potentiometer. This divided voltage is fed to negative input of LM358 comparator amplifier. At the positive input there is a previously set fixed voltage (for example 2.5 V).

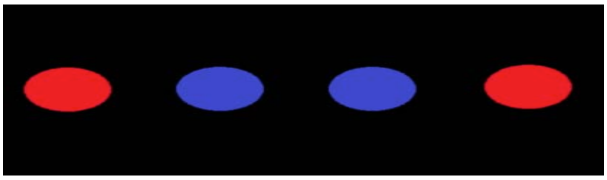

Figure 6. Android display on robot

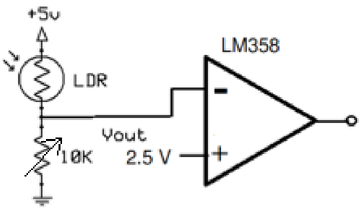

Figure 7. Voltage comparison

LM358 gives a high output (logic 1) when the voltage of its positive terminal is greater than the voltage at the negative terminal. When any spot is blue at android display, the Vout is less than 2.5 and therefore the comparator output is high. When a spot is red, the output is low.

The 4 outputs of the comparator are taken as input in the microcontroller and it is programmed accordingly. When the four input bits change, PWM signal is generated at the microcontroller output. As we have used 4 bits, there can be total 16 combinations. In our project, we used 5 commands for different directions.

The L298 motor driver controls the speed and direction of the two DC motors at the wheels. It takes PWM input at Enable A and Enable B for motor 1 and motor 2 respectively. In1 and In2 control the direction of motor 1. If In1 is set to 0 and In2 is set to 1, the motor rotates is one particular direction. Reversing the bits changes the rotation direction of the motor.

Table 1. Commands, Displays and Motor Directions

Command	Display	Motor Directions
Stop		Both motors stopped
Forward		Motor1= Clockwise Motor2=Clockwise
Backward		Motor1= Anti-Clockwise Motor2=Anti-Clockwise
Right		Motor1= Clockwise Motor2 Stopped
Left		Motor2= Clockwise Motor1 Stopped

E. *PCB Layout:*

The PCB is designed in single layer with trace width 40 and pad to pad distance of 30. Due to some complexity of design we had to use some wire jumps.

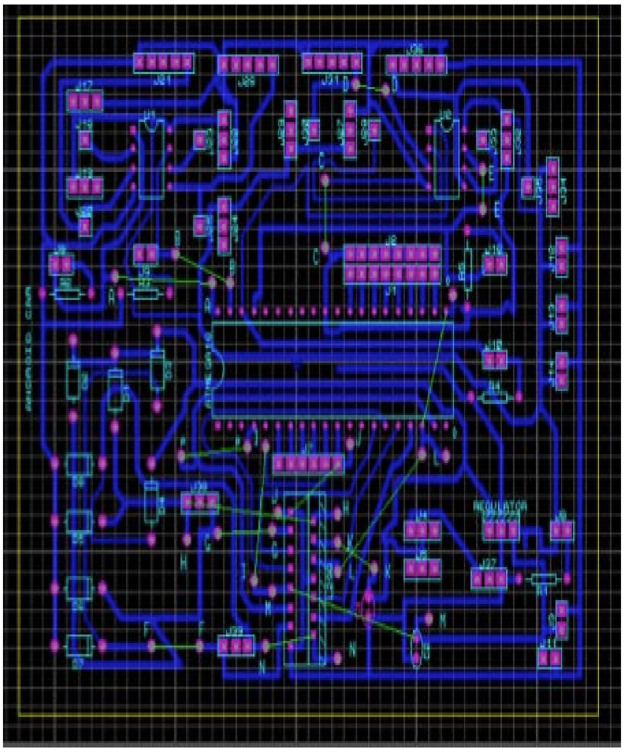

Figure 8. PCB Layout

By using these we have implemented the robot practically.

Figure 9. Actual Robot Body

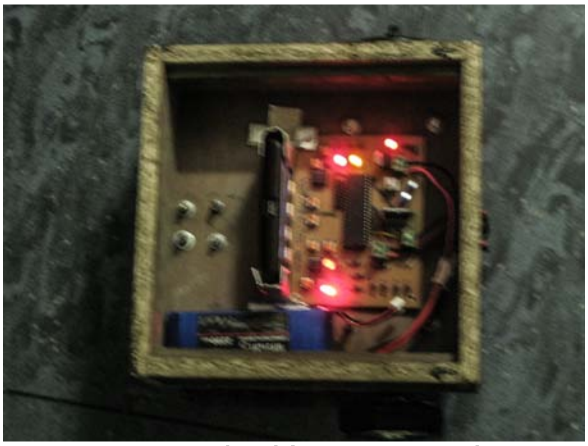

Figure 10. Robot while receiving command

VI. CONCLUSION AND FUTURE RESEARCH

Although this research is still in an early stage of development, it has already proven to succeed in several of its goals. We have implemented five basic tasks which can be done by the robot taking command from an android app which uses GCM technology. This meaningful two way communication between the android device and the robot allows a non-expert to interact with this robot very easily.

Since this Mobile robot is not developed with a professional team of developers and the GCM technology is quite new technology, there are some scopes to be improved. However currently it is operational and usable and will be developed in the near future for more accuracy and to make it more user-friendly. By implementing the Android platform with a microcontroller, we let the microcontroller use the vast functionality of android platform. By adjusting a proper measurement we can implement android map with a microcontroller. That will enable us to use the GPS and send the robot to a specific location by choosing the path from the map. Online Map and voice recognition service can be implemented as well. Use of high quality batteries and other equipments will make this robot more effective and look more attractive.

REFERENCES

[1] Tarantola, Andrew (June 27, 2010). "Google’s New Cloud Messaging System Does More for Less". Gizmodo. Retrieved August 10, 2012.
[2] GCM Developers portal by Google, http://developer.android.com/google/gcm/gcm.html
[3] Sugoog Shon, Protocol Implementations for Web Based Control Systems, International Journal of Control, Automation, and Systems, vol. 3, no. 1, pp. 122-129, March 2005.
[4] Web based control system design & analysis, 02721708/04/$20.00©2004IEEE, IEEE Control Systems Magazine, June 2004.
[5] M. pelko, K. zagar , L. zambon, A. J. Green, Canone- a highly-interactive web-based control system interface, proceedings of icalepcs07, knoxville, tennessee, usa.
[6] E. Cardozo, E. Guimaraes, L. Rocha, R. Souza, F. Paolieri, and F. Pinho, “A Platform for Networked Robotics,” IEEE/RSJ International Conference on Intelligent Robotics and Systems, Taipei, Taiwan, 2010.
[7] Z. Wang, L. Liu, and M. Zhou. “Protocols and Applications of Ad-hoc Robot Wireless Communication Networks: An Overview,” The International Journal of Intelligent Control Systems, vol. 10(4), pp. 296-303, 2005.
[8] T. Fong, I. Nourbakhsh, and K. Dautenhahn, “A Survey of Socially Interactive Robots,” Robotics and Autonomous Systems, vol. 43, pp.143-166, 2003.
[9] van Delden, S.; Whigham, A., "A bluetooth-based architecture for android communication with an articulated robot," *Collaboration Technologies and Systems (CTS), 2012 International Conference on* , vol., no., pp.104,108, 21-25 May 2012.
[10] Z. Mednieks, “Programming Android,” O’Reilly Media, ISBN:1449389697, 2011.
[11] R. Meier, “Professional Android 2 Application Development,” Wrox Publishers, Second Edition, ISBN: 0470565527, 2010.

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Analysis of Normal and Infected Bio-cell by Using Dual Nanoprobe

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***Abstract*— Knowledge of nanoprobe based bio-cell analysis method can be used to diagnostically difference between healthy and infected bio-cells. This method is made possible by using nanotechnology, a new field of science that provide a technology for human to interact with nanoscale life form organism specially cell. The electrical behaviors of healthy and infected cells are different. This paper analyses yeast cell, liver cell, and blood cell in both healthy and infected conditions to observe the differences in electrical behaviors. A dual nanoprobe is used for supplying electrical power from source to bio-cell. The voltage can be applied by two ways across the cells. One is penetrating the cell wall and another is keeping the nanoprobe in closed contact with the cell membrane. After simulation the current was measured about 2.7 times larger for liver tumor cell than healthy liver cell including cell membrane. The current flow through the healthy cell is 1.9nA whereas the current flow through a dead cell is 34pA. It is expected due to the conductivity of cytoplasm of healthy cell is greater than that of dead cell. The current is measured for a leukemia affected cell is 21.2nA. It is 2% less than the current for a white blood cell.**

Keywords: Dual nanoprobe, Leukemia affected WBC, Yeast cell, Liver cell, Electrical properties, Conductivity.

I. INTRODUCTION

Nanoprobe based bio-cell analysis method is a very newer form of method. For example single cells electrical characterizations using nanoprobe via ESEM-Nano-manipulator System, one of the cell viability detection methods, was invented by a group of researchers in 2009 [1].

This method introduces the cell analysis in nanotechnology, a new field of science. In the analytical procedure, a single cell is analyzed by measuring the current through the cell by the application of a dc voltage using dual nanoprobe. Penetration (see in Figure 1) and without penetration (only contact with cell membrane; see in Figure 2) of the nanoprobe into the cell are the two processes for applying voltages across the cell. It is best to simulate the method first since the method is still new. For this purpose, ABAQUS 6.10 CAE, powerful finite element software has been used to simulate the experimental method.

It is observed that, a bio-cell shows various types of electrical behavior. Among them two behavior are taken into account for cell analysis. One is, conductivity of cytoplasm of cancerous or infected cell is lower than healthy cell. Because when the cell is dead or cancerous, the amount of ions becomes lower than healthy cell [2]. Another behavior is the conductivity of cell including membrane shows higher conductivity in case of cancerous or infected cell than the healthy one [3, 4], since the permeability of membrane increases, when the cell becomes cancerous. As a result more ions can flow into the cell [3, 5]. Moreover cancer cells have altered membrane composition and membrane permeability, which results in the movement of potassium, magnesium and calcium out of the cell and the accumulation of sodium and water into the cell [3, 5], results the flow of more ions into the cell.

The conventional method of cell viability and cancer detection is done by using chemical substance [1]. Colorimetric or florescent dyes are used for cell viability detection. The limitation of this method is the lack of capability to produce instantaneous and quantitative result but nanoprobe based cell analysis method is much better in terms of producing instantaneous and quantitative result [6].

Bone marrow aspiration is a conventional type of biopsy used to diagnose leukemia. In open biopsy, the bone is taken out and stitches are given to the patient. For this, the patient has to stay back in the hospital for few days. Patient may experience bleeding after the procedure. The nanoprobe based testing method may be better in this case for leukemia detection.

In this paper, section-I was discussed about theoretical background, analytical procedure and the advantages of this nanoprobe based novel analysis process in comparison to the conventional method; in section-II process of implementation is discussed; the simulation procedure is discussed in section-III and section-IV, V and VI are discussed about result and analysis, discussions and conclusions respectively. Here the result and analysis section represents the simulation output and compares the output data with previous data.

II. METHODOLOGY

A dual nanoprobe is needed to supply an electrical power from source to the bio-cell. The one end of nanoprobe has to be connected with 2V dc source shown in Figure 1 and the another end of nanoprobe has to be penetrated into the cell wall for both yeast and blood cell or have to be in closed contact with the cell membrane for liver cell shown in Figure 2.

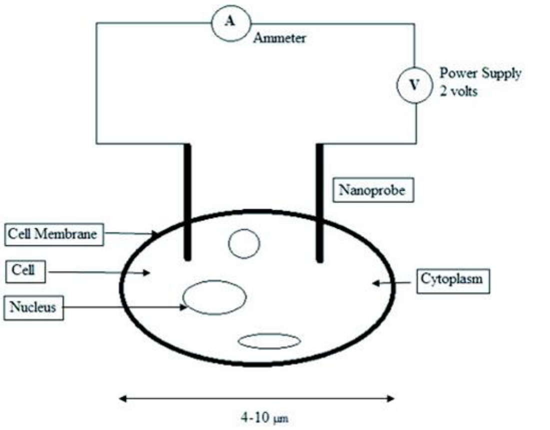

Figure 1. Schematic diagram on penetration of duel nanoprobe into a cell [1].

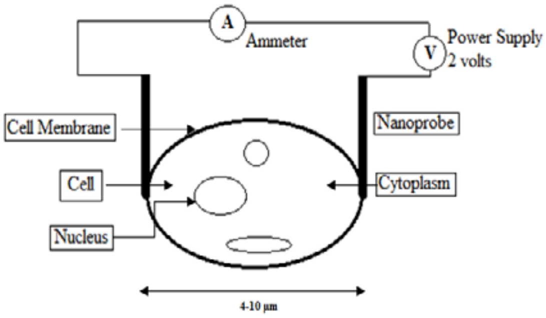

Figure 2. Schematic diagram of edge contact of dual nanoprobe into a cell.

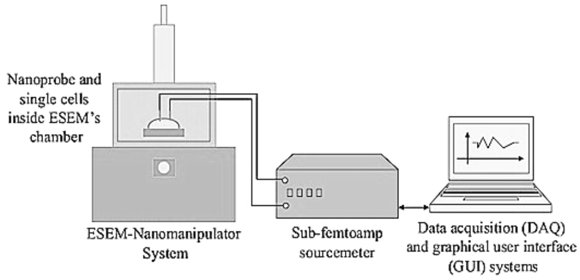

Figure 3. Experimental setup [1].

The cell can be analysed via ESEM-Nano-manipulator System. The output current is measured using sub-femto ampere source meter. This analog current is converted into digital by data acquisition (DAQ) and this digital data is then represented by graphical user interface (GUI) systems shown in Figure 3.

III. SIMULATION PROCEDURE

ABAQUS 6.10 computer aided engineering (CAE) software that provides strong platform to design and simulate any type. This software has the ability to use in nanoscale level simulation. Figure 4 shows the simulation setup designed in ABAQUS platform that shows two nanoprobes touching the cell-membrane. The properties of basement and the probe martials has to be the same.

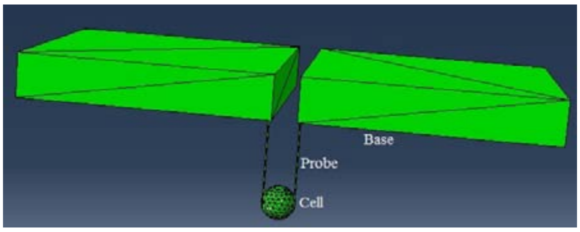

Figure 4. Simulation Setup.

Otherwise result will be varied due to dissimilar materials junction's conductivity of the system. The steps of simulation procedures are represented by a simple flow chart shown in Figure 5.

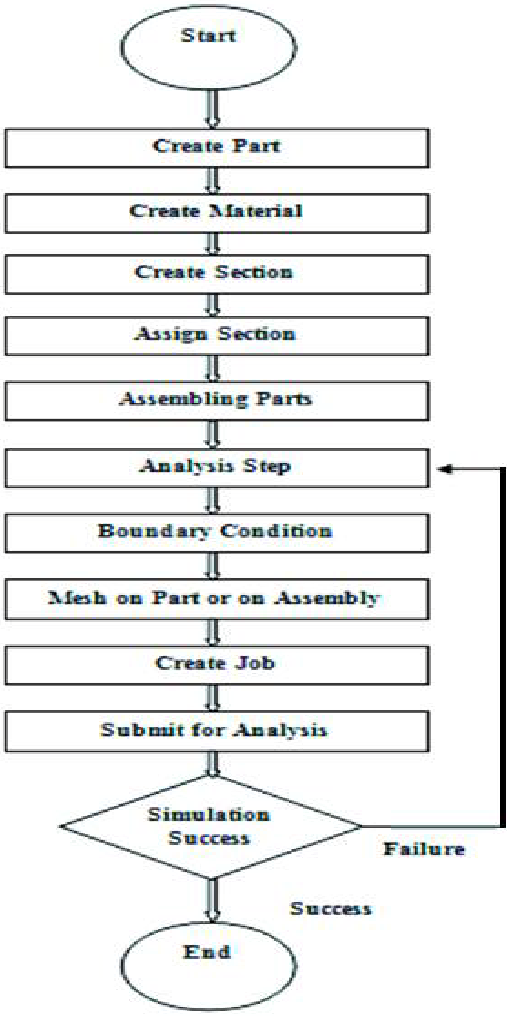

Figure 5. Flowchart of simulation procedure.

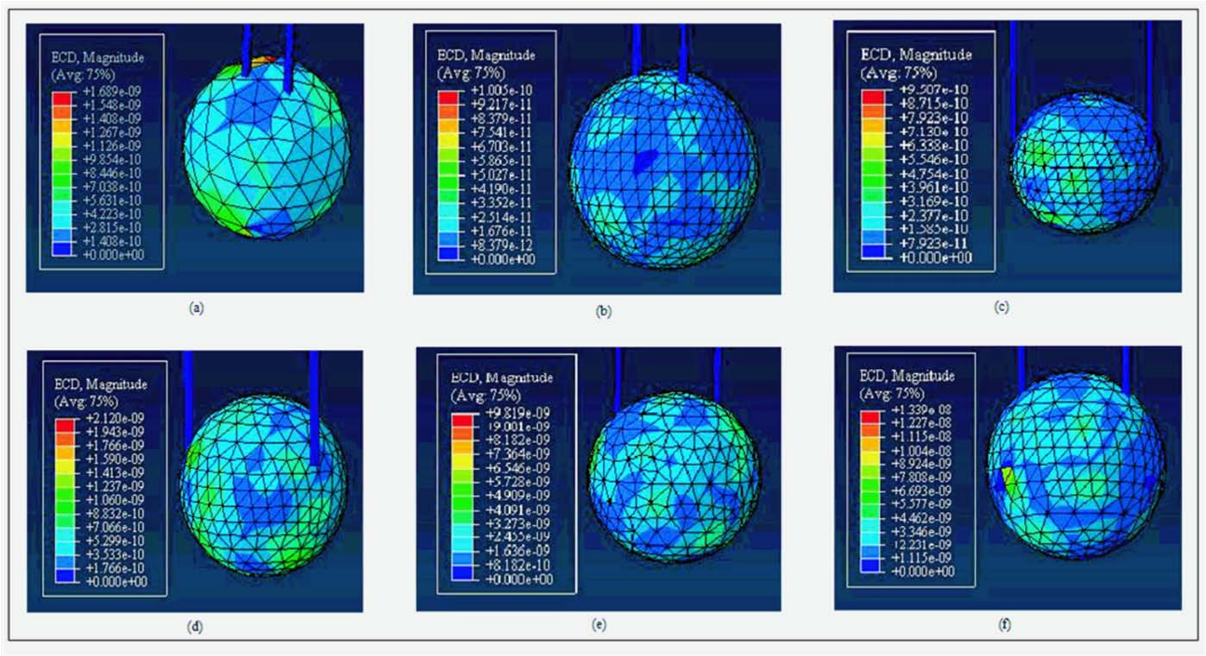

Figure 6. ECD value for different cell model. (a) healthy yeast cell, (b) dead yeast cell, (c) liver cell, (d) liver tumor cell, (e) white blood cell, (f) leukaemia affected WBC cell.

The major part of the study is to characterize the nanoprobe based on type of material and cross section. Resistance is the main factor of the probe. Many types of materials can be used to construct the probe but that material can provide the best result which low resistance and no loading effect on load (cell). The size of nanoprobe used is 15µm long and 200nm ×200nm cross-section area. The probe is made of gold. Cell characterization and simulation is the basic part of this study. A cell model is designed as close as real cell. Different mechanical and electrical properties are given to approach a model, which is seemed to be a real cell.

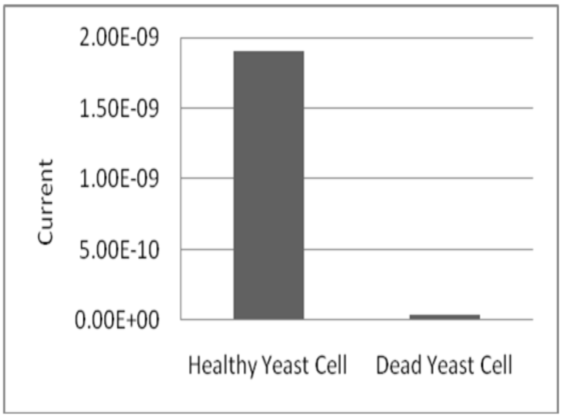

Figure 7. Current in a healthy and dead yeast cells.

IV. Result and Analysis

The current flows through the cells are represented by electrical current density (ECD). The output current is found by the product of average cross-section area of the cell and average electric current density (ECD) which is given by the equation (1).

$$I = J_{ECD} \times A \quad (1)$$

Where I is the current, J_{ECD} is the electric current density in the cell and A is the cross-sectional area of the cell under the probes. Figure 6 represents the electric current density for different cell model. The resultant J_{ECD} are 268pA/µm^2, 4.80pA/µm^2, 162.6pA/µm^2, 443.4pA/µm^2, 3037.0pA/µm^2 and 3000pA/µm^2 for healthy cell, dead cell, normal liver cell, liver tumour cell, white blood cell and leukaemia affected white blood cells, respectively.

By using resultant ECD magnitudes, a current that flows through the cell can be drawn according to the equation (1). Simulation outputs of the calculated current values become 1.9nA and 34pA for healthy and dead yeast cells shown in Figure 7. It is obvious that leaving yeast cell is highly conductive than a dead cell.

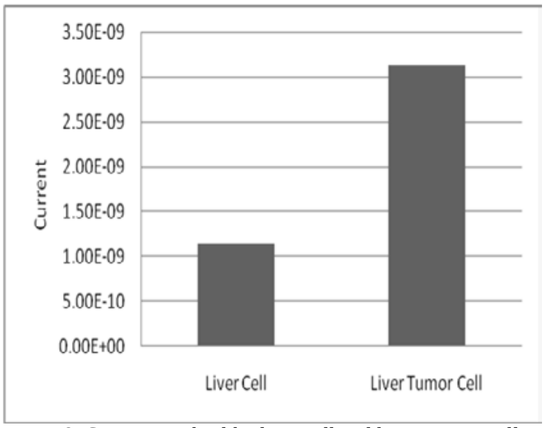

Figure 8. Current in a healthy liver cell and liver tumour cell.

On the other hand a healthy liver cell conducted current 1.1486nA and the liver cell with tumour conducted current 3.1327nA which is shown in Figure 8. The current level of a liver tumour cell increases due to increasing the ions in the liver tumour cell.

Figure 9 represents the currents for healthy WBC and the leukaemia affected WBC. The healthy WBC conducted the current 21.457nA and the leukaemia affected WBC conducted the current 21.2 nA, respectively. In this case leukaemia reduces the conductivity of the WBC.

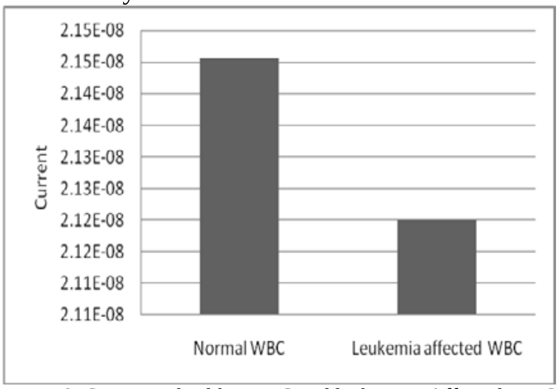

Figure 9. Current in healthy WBC and leukaemia Affected WBC.

The main approach of this study is to segregate the current value in between normal and infected cell. The current flow through the healthy cell is 1.9nA and dead cell is 34pA for yeast cell, which is expected. This is because; the conductivity of cytoplasm for healthy cell is greater than that of a dead cell. On the other hand the current was measured about 2.7 times larger for liver tumour cell than that of healthy liver cell. For measuring the current against WBC should be more precise, because the current as well as conductivity difference is only 2%.

Table 1, Results comparison between experimental and simulation

Types of Cell	Experimental result with 2V supply and with (probe + base + cable) resistance 1kΩ	Previous simulation result with 2V supply and with (probe +base) resistance 37.46Ω	Present simulation result with 2V supply and (probe + base) resistance 107.2Ω
Healthy (yeast ell)	262pA current	54mA current	1.9nA current
Dead (yeast ell)	2pA current	0A current	34pA current

Table 1 and Table 2 show the comparison among experimental data, previous simulation and present simulation done in this analysis. The probe gap was taken 1.46μm for penetration. The current values for present simulation are high compare to experimental result because resistance is low as there is only probe and base is considered.

Table 2, Results comparison between experimental and simulation for Yeast cell.

Properties	Experimental Result	Previous Simulation	Present Simulation
Resistance ,Ω	1k	37.46	107.2
Sensitivity, mA/V	1	27.6	9.3
Voltage, volt	2	2	2

V. DISCUSSIONS

The test may give perfect result if the cells are tested in same environment, keeping same probe gap and same depth of penetration because the conductivity between probes across the cell may vary with probe gap, depth of penetration and environmental condition surrounding the cell. More than one sample of blood cell having WBC should be tested for better confirmation of the presents of immature WBC as the conductivity difference between normal WBC and leukemia affected WBC is only 2%. As well as for other cells detections, same tasks should be taken.

VI. CONCLUSIONS

This nanoporbe based analytical process may have better opportunities, when it will be practically implemented. For detection of healthy, death and cancerous cells (leukemia), this process may give instantaneous and better results. Solution of process of analysis provides more accurate result then conventional ways as nanoprobe deals with any change of result in nanoscale level. A long way left to completion the research in this method. This project is only a small portion of a big idea. Simulation is one of the universal approach that researcher use in their research. This novel nanoprobe based detection process may introduce with versatile opportunities for researchers and a revolutionary change can occur in medical science specially in the section of detection, analysis and treatment of diseases.

REFERENCES

[1] Abdul Hafiz Bin Mat Sulaiman, "Study of Novel Method of Cell Viability Detection Based on Electrical Measurement

Using Dual Nanoprobe", *Enabling Science and Nanotechnology (ESciNano), 2012*, p. 1 - 2.

[2] S. Ingebrandt, G. Wrobel, S. Eick, S. Schäfer and A. Offenhäusser, "Probing the Adhesion and Viability of Individual Cells with Field-Effect Transistors", *Solid-State Sensors, Actuators and Microsystems Conference, 2007. TRANSDUCERS 2007. International*, p. 803 - 806.

[3] Steve Haltiwanger M.D., C.C.N, "The Electrical Properties of Cancer Cell. h ttp://www.royalrife.com/haltiwanger1.pdf.

[4] Ranjit Kumar Sahu a,1, Udi Zelig b,1, Mahmoud Huleihel c, Nir Brosh d, Marina Talyshinsky c, Miriam Ben-Harosh d,

[5] Shaul Mordechai, Jossi Kapelushnik d,"Continuous monitoring of WBC (biochemistry) in an adult leukemia patient using advanced *FTIR*-spectroscopy", June 2005. [Online] Available : http://www.sciencedirect.com.

[6] Tanay Tandon, " Novel Detection of Hematological Cancer: A proposed conductivity based analysis for early Leukemia cell identification", http://www.usc.edu/CSSF/Current/Panels/S05.html#S0531.

[7] John Douglas Stenson, " Investigating the Mechanical Properties of Yeast Cells", May 2008. [Online] Available : http://etheses.bham.ac.uk/304/1/Stenson09PhD.pdf.

[8] Seeger PG, Wolz S. "Successful Biological Control of Cancer: By Combat Against the Causes. Gesamtherstellung: Neuwieder Verlagsgesellschaft mbH", 1990, [Online] Available : www.buergerwelle.de/assets/files/grn/ElecPropCancer.wiz.

[9] Damijan Miklavcˇ ICˇ, Natas ˇ A Pavs ˇ Elj, Francis X. Hart, " Electric Properties of Tissues". [Online] Available : http://www.google.com/electirc properties of tissues.

[10] Dieter Haemmerich, S T Staelin, J Z Tsai, S Tungjitkusolmun, D M Mahvi and J G Webster ," In vivo electrical conductivity of hepatic tumours.", *Physiological Measurement*, Volume 24, Number 2, 7 February 2003.

[11] Efstathios Karathanasis, PhD, Sankararaman Suryanarayanan, PhD, MBA, Sri R. Balusu, Kathleen McNeeley,MS, Ioannis Sechopoulos, PhD, Andrew Karellas, PhD2, Ananth V. Annapragada, PhD, Ravi V. Bellamkonda, PhD,"Imaging Nanoprobe for Prediction of Outcome of Nanoparticle Chemotherapy by Using Mammography.", *Radiology*,February 2009.

[12] Beckman Coulter Inc. (2010). Automated viability soultions Vi-Cell Series Cell Viability Analyzers. U.S.A : BMR.

[13] Foster KR, Schepps JL. "Dielectric properties of tumor and normal tissues at radio through microwave frequencies.", *J Microwave Power* 1981;16:107-119.

[14] Jian Chen, Mohamed Abdelgawad, Liming Yu, Nika Shakiba1, Wei-Yin Chien1, Zhe Lu,William R Geddie, Michael A S Jewett and Yu Sun, "Electro-deformation for single cell mechanical characterization", *JOURNAL OF MICROMECHANICS AND MICROENGINEERING,* 28 April 2011.

[15] Hans G. Keitel, H. Berman, H. Jones and E. Maclachlan, " The Chemical Composition of Normal Human Red Blood Cells, including Variability among Centrifuged Cells", [Online] Available: bloodjournal.hematologylibrary.org.

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Development of Asynchronous Replication Model for Heterogeneous Environment

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Abstract **– Replication is a very useful technique in distributed systems, grid community and clustering systems. Now a days a lot of algorithm development has been focused for safe replication services. Replication can improve performance, reliability, portability of entire database. In this paper, a persistent layer has been developed and proposed which supports heterogeneous system. The persistent layer work on asynchronous model, hence it is known as asynchronous replication. The implementation of this algorithm has built on Java based technology thus it is very easy to deploy any OS without hassle and configurable files makes the whole system easy to maintain and cost effective. The experimental servers (both main server and replication server) used both windows and Linux OS's. Finally some experiments have been carried out by different data taken from references. The results show that, the proposed model outperforms all available replication models.**

Key Words: *Data grid, data replication, asynchronous, persistence layer, heterogeneous environment, multi-threading technique.*

I. INTRODUCTION

In large distributed database environment, data grid mainly deals with large computational problems. It can support to manage geographically distributed resources for large-scale data-intensive applications that may generate huge scientific data sets. For such large-scale systems scientists need to managing the massive amounts of largely geographically distributed data (Abdullah et al, 2008). The effective management of these large amount of information shared and distributed is becoming a topic of scientific research for commercial applications. In a nutshell, data replication is very useful for distributed database systems. (Sriram and Cliff, 2010; Li and Shen, 2009; Lei et al., 2008). Over the decade, many researchers including Ibison(2010), Pucciani et al, (2010), Tanga et al. (2010), Sato et al. (2009), Elghirani et al. (2007), Boyera and Hura (2005) and Ma et al. (2004) has extensively worked to propose different replication models. Prior research shows that data replication in heterogeneous system is an issue of research and the study was initiated.

Replication of data is the process of keeping multiple copies of data also known as replica. This replications can improve performance as; i) reduce latency by using nearby replicas and avoiding remote access; ii) accelerate performance by using service of multiple computers simultaneously (Wahid et al. , 2007; Sashi and Thanamani, 2011). In distributed systems, replication, consistency and data integrity are used as correctness criteria (Osrael et al. 2007).

There are various applications of data replication in banks, insurance, industries to protect and secure their data. In replication data are replicated in terms of duplication in different servers. In enterprise software they use a persistence layer which persists in current objects, which help the application avoid fault tolerance. Current replication systems have the following drawbacks:

- Replication solely depends on main server.
- System modifications of the replication system usually force to halt/stop the system for a routine of time.
- Any major problem including shutting down of main server, force entire system to stop working, especially in a DB driven system.

Our research has focused for developing a Persistence Layer for Asynchronous Replication (PLAR) in data grid environment which also supports heterogeneous systems. Our proposed layer uses a multi-threaded application which also provides an interface between the database and the main system similar to Hashem et Al., 2012. Development of persistent layer was primarily introduced by Hashem et, al 2012 on a synchronous replication model.

II. RELATED WORK

Replication of data is a process that improves the performance of accessing of data in a large distributed system. By using this technique, a read or write (object request) will be accessed from multiple locations in a network environment (in a LAN or in a WAN). For example, the financial instruments price will be read and updated from around the world (Noraziah et al., 2009; Chidambaram et al., 2008; Gu et al., 2006).

A. Data Grid Solution

Data are typically replicated for improvement of file access time and data availability in a data grid. The resource of many computers lying in large geographic locations and organizations are utilized to solve problems that requires huge computations in the system (Lei et al. 2008). The true objective of data replication is to create multiple copies of files in different location to increase their data availability (M.Mat Deris et al., 2004; Atakan, 2009; Khanli et al., 2011). Shena and Zhu (2009) proposed a file replication scheme having low overhead, where it creates replicas among physically close nodes. Atakan (2009) studied for dynamic file replication for real-time file access in data grids. Data replication is used in data grid to enhance data availability and fault tolerance (Bsoul et al., 2010). However, it is required to take care of storage of nodes of a data grid while taking decision of replication.

Replication of data will improve the job response time and data availability. (Lei et al., 2008) has proposed an algorithm which is line optimizer and that can minimize the data missing rate in order to maximize the data availability. In (Chang et al., 2007), proposed a job scheduling policy, named ‘Hierarchical Cluster Scheduling’, and a dynamic replication model, called ‘Hierarchical Replication Strategy’ which was implemented to improve access efficiencies in cluster grid.

Persistent layer based synchronous replication model as proposed in 2012 (Hashem et. al., 2011) which was developed using multithreaded technique. The persistent layer was presented as an engine which was responsible for scheduling and coping data from main server to replication server.

B. Persistence Layer

For a collaborative computing environment, the collaborative applications require a simple and transparent persistence middleware to deal with complex data access. Persistence layer is used in load balancing systems and in data dictionary. It is very important in parallel and distributed system. Qiao et al. (206) described two such frameworks for parallel and distributed systems. Wang et al. (2005) proposed a persistence mechanism called Tree-Structured Persistence Server. It allowed states of collaborative applications to store in a tree fashion beside tables.

C. Heterogeneous system

In recent days, heterogeneous systems became important as a major high-performance computing platform in distributed and parallel system. A Distributed Heterogeneous Computing (DHC) is a collection of autonomous dissimilar computing machines that are linked by a network and synchronized by software which functions like a single powerful computing facility. A DHC system has advantages over homogeneous computing (Boyera and Hura, 2005).

Chen et al. (2008) have proposed a model based on mark speed for heterogeneous computing. Chi et al. (2006) proposed App.Net.P2P architecture for implementing effective content delivery on P2P networks for heterogeneous system. The objective of App.Net.P2P is to allow delivering intermediate objects to other peers as well as the final presentation.

D. Transactions

A group that contains a set of tasks of any application that collects or manipulates data is called a Transaction. It means any two users should not modify the same data item simultaneously. In any transaction all statements are treated as a group of work. Any failure of any statement treated as a group to be failed. If all the statement succeed the transaction is treated as a succeed transaction (Dewald, 2011; Poddar, 2003).

E. Replication in SQL server

Microsoft has designed SQL as a Relational Database Management System (RDBMS) for the enterprise environments. This runs on Transact-SQL, a programming set extension from Sybase and Microsoft. Gutzait (2007) has described SQL, where the architecture shows that the database is replicated with a common structure and the changes are replicated to the subscribers with fewer publications. All the database servers are connected with the main server and the main database also connected with replication servers. SQL server replication is show in Figure 1 below:

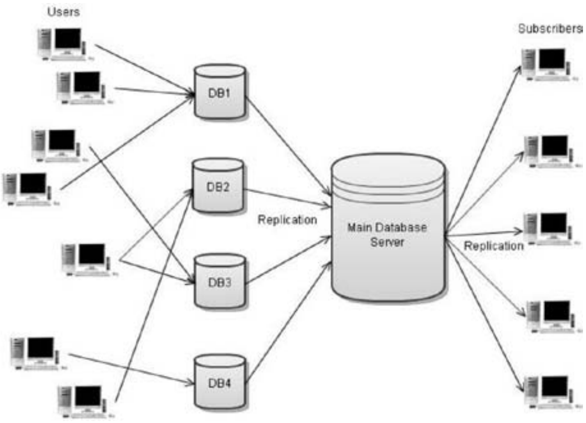

Figure 1: SQL server replication architecture

Ibison (2010) shows a comparison of SQL server replication times between Merge and Transactional insert and Synchronization as in Table-1. The data in table shows that for merge, inserts, it takes significantly longer than transactional and snapshot.

Table 1: Basic Comparison of SQL server replication time between transaction and merge insert

Rows	Snapshot Inserts	Sync Time	Transactional Insert	Sync Time	Merge Insert	Sync time
100	0	4	0	0	1	2
500	1	4	1	1	3	6
1000	2	5	2	1	5	12
5000	6	7	7	2	21	42
10000	13	26	26	5	42	96

III. PROPOSED MODEL DESCRIPTION

In our proposed system, it modifies the persistence layer to adopt multi-processing by the use of multi-threading. Here the persistence layer will be connected with different database servers and from those, one will be the main server and that will take care Create, Read, Update and Delete with the entire system and the several others will be known as the replicated server. A single high priority system thread will keep the consistent connection with the main server. Other than that another thread will be responsible for replication. It will sense the CRUD operation in main server, and reflect the changes to replication servers.

The thread is defined as the single stream of execution within a process. Processes are programs that execute within its own address space. For each replication server the persistence layer will create a queue. . There will be two threads, one will be connected to main server with highest priority and another will be less priority which will be responsible for replication. Both threads will be manage and maintain from persistent layer. Since the main server is maintained with a higher priority thread, thus the data should be saved or deleted or modified immediately with high priority. Consequently, all the data transactions will be in queue and it makes its own copy and does the transaction with that thread.

A. Programming implementation

Our proposed model has been implemented in programming by using Java Language. NetBean 7.0 has been used to write the source code of the persistence layer. NetBean has been chosen because of its comfortable Integrated Development Environments. NetBean is developed and maintained by Sun Microsystems. NetBean has its own file format to maintain the source code. MySQL, SQL Server and MS Access was used as the database. A free tool called SQLyog was used to implement the SQL Query testing. For connecting SQL with Java, we have used Java-MySQL connector. For MSSQL, Java SQL Server connector API was used.

B. Hardware and software components

Our proposed model has been implemented with software, which required some minimum hardware specifications. The model was demonstrated with across three prototype replication servers as shown in Figures 2 and 3. Each server or node was connected to other by using fast Ethernet switch. Theoretically, all the replication servers and the main server had to be connected to each other logically. The hardware specification that was used to demonstrate our model is shown in the Table 2. Each of the computer involved in this model used different OS environment. Table 3 shows the tools used for the system development and implementation.

Table 3: Server components specification

Hardware	Specification
Processor	Intel ® Core 2 Quad Q9650 @3.00 GHz
Cache	2048 MB
Primary Memory	4.00 Gigabyte
Chip Set	ATI Radeon HD 3450-Dell OPtiplex
Hard Disk	500 Gibagyte
Network Card	Intel® 82567 LM-3 Gigabit Network

C. Proposed Model Environment

The functionality offered by our proposed model was for heterogeneous systems. For the implementation of our model we have tested different server under a single local area network (LAN). Two different experiments have been done at implementation stage. In our first experiment SQL server was used as the main server, and in second experiment, MySQL server was used in Linux OS environment as the main server.

Table 2: System Development tools specification

System development Software	Specification
Windown XP	TM Professional
Ubuntu	Version 10.0.4
SQL Server	Version 2008
MS Access	Version 2007
MySQL server	Version 5.0.89
NetBeans IDE	Version 6.9.1
Java	SE (Jdk 6.0)
SQLyog	Community edition 8.53
Wine	Version 1.14

i. Experiment No.1

In our first experiment, replication in heterogeneous systems has been done with one main server and three replication server. Here our persistence layer has been used to connect entire servers. Here SQL server was the main server and it was connected with the persistence layer. Our first replication server was MS Access, MySQL in Windows OS was the second replication server and MySQL in Linux OS was the third replication server, where all these server were connected as show in Figure 2. The host name and IP address for each server are shown in Table 4. Main server A and replications servers B,C and D was assigned the IP's as 172.1.1.222, 172.1.1.223, 172.1.1.224, 172.1.1.225 respectively.

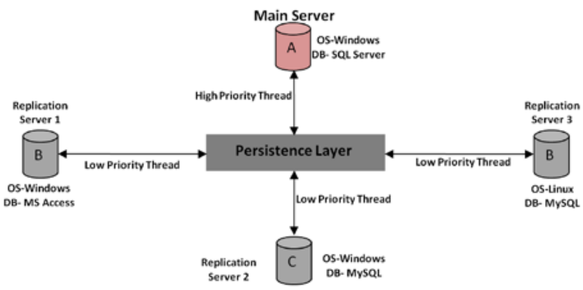

Figure 2: Replication on heterogeneous system with 3 replication server and SQL server

Table 4: The Local IP address for each server based on SQL

Server	Host Name	IP Address	O/S	Software
A	Main Server	171.1.1.222	Windows XP	SQL Server
B	Rep. Server 1	171.1.1.223	Windows XP	MS Access
C	Rep. Server 2	171.1.1.224	Windows XP	MySQL
D	Rep. Server 3	171.1.1.225	Linux (Ubuntu)	MySQL

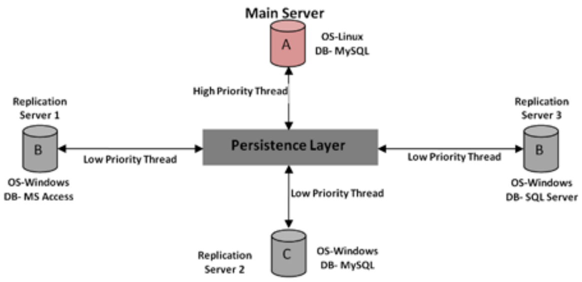

Figure 3: Replication on heterogeneous system with three replication server and MySQL server.

ii. Experiment No.2

For our second experiment, heterogeneous replication was done with same one main server and three replication servers. Like our first experiment, the entire servers are connected with the persistence layers. Only the difference was, MySQL in Linux OS was the main server which was connected with the persistence layer with high priority thread and MS Access was the first replication server, MySQL in Windows OS was the second and SQL server was the third replication server with low priority threads as shown in Figure 3. The IP addresses of the host and the other servers are shown in Table 5.

Table 5: IP address for servers based on MySQL server.

Server	Host Name	IP Address	O/S	Software
A	Main Server	171.1.1.222	Linux (Ubuntu)	MySQL
B	Rep. Server 1	171.1.1.223	Windows XP	MS Access
C	Rep. Server 2	171.1.1.224	Windows XP	MySQL
D	Rep. Server 3	171.1.1.225	Windows XP	SQL Server

Table 6: Transactional insertion time between SQL and Proposed replication server 1 & 2

No of rows	SQL Server Rep.	Proposed Rep. Server1	Proposed Rep. Server2
100	0	0.375	5.129
500	1	0.459	5.385
1000	2	0.689	5.479
5000	7	2.421	6.485
10000	26	4.525	7.508

Table 7: Transactional sync. time between SQL and Proposed replication server 1 & 2

No of rows	SQL Server Rep.	Proposed Rep. Server1	Proposed Rep. Server2
100	0	0.375	5.129
500	1	0.459	5.385
1000	1	0.689	5.479
5000	2	2.421	6.485
10000	5	4.754	7.508

IV. OBSEVATIONS & RESULTS

Our asynchronous replication in persistence layer has been compared with other replication processes in terms of replication time for transactional insert. From our experiment, Table 6 shows the comparative time between SQL Server and our proposed model's replication server 1 and server 2 for insertion. In first experiment, server 1 was the main server and for the second experiment server 2 was the main server. The result shows that, for 1000 rows of data insertion, SQL server takes 2.04 second, whereas our proposed model's replication server 1 and server 2 takes 0.389 and 5.179 seconds respectively. For 5000 row of data insertion SQL server takes 7 seconds where our proposed model's replication server 1 and server 2 takes 2.121 and 6.213 seconds respectively. Similarly for 10000 rows of data insertion, SQL server and our proposed replication server 1 and server 2 took 26 second, 4.754 and 7.508 seconds respectively. Table 7 shows the result of comparative time between SQL server and our proposed replication server 1 and server 2.

The experiment result demonstrates that in 1000 rows of data replication time, our proposed replication server 1 and server 2 took 0.439 and 5.320 second respectively, and for 5000 rows of data sync. time, SQL server took 2 seconds were our proposed replication server 1 and server 2 took 2.421 and 6.485 seconds respectively. Similarly for 10000 rows of data sync. Time; SQL server have taken 5 second where our proposed replication server 1 and server 2 took 4.321 and 7.023 seconds respectively. Here the total replication time (R_T) is calculated by using the equation (1) (Beg et al., 2011):

$$R_T = \sum (T_T + S_T) \qquad (1)$$

Where, R_T is the replication time, T_T represent transactional time and S_T represent replication time.

So, for 10000 rows of data insertion in SQL server, (Table 1)

$$R_T = (26+5)s$$
$$= 31s$$

for 10000 rows of data insertion in Proposed replication server (server 1), (Table 6)

$$RT = (4.525+0)s$$
$$= 4.525s$$

[S_T=0 because in our proposed model T_T and S_T done at the same time

Replication time required for the proposed replication model's replication server1 and server 2 and SQL server have been compared.

V. DISCUSSIONS ON RESULT

It can be observed from the replication comparison table that our proposed replication server need lower replication time than the SQL server replication. As the number of rows goes higher, SQL server's replication time gets higher in comparison with our proposed model.

The result shows that our proposed model performs outstanding than SQL server for transactional insertion in comparison to SQL server replication in time per seconds. The motivation to compare the result of our proposed replication with SQL server replication is the transactional replications that can alter the use of several trigger, similar to the proposed strategy, as the algorithm can perform rollback command from the persistence layer which can alter the result. From the abovementioned result and execution point of view, it can be found that our proposed model is highly acceptable.

VI. CONCLUSION

Now a day, in the grid community and clustering system, a lot of scope has been focused on providing efficient and safe replication management services by designing of algorithms and systems. The data grid and grid computing system has to process and manage such large amount of distributed data. Many organizations use replication for different purposes. In our research, a new model has been proposed for managing data replication in the heterogeneous systems. This model can provide several advantages for enterprise application, a more secure and reliable data transmission. A major goal of our research is to make database replication much easier to handle. Thus, we made it vastly configurable and the entire architecture, service oriented. We used current technology trends and the replication is done from the persistence layer. Since the persistence layer is the part of a software engine and uses the latest customizable fourth generation language like Java, so a new era can move forward related to networking as well as database programming. For time being our system supports only 3 types of database, supporting all the existing databases can be another important improvement of our proposed system. Our developed prototype tool has fewer fields to insert data into different master and replication table. To make this more user friendly, various input can be introduced. In future more work need to be done to improve the system's capability.

REFERENCES

[1] Atakan D (2009). A study on performance of dynamic file replication algorithms for real-time file access in Data Grids. Future Generation Computer Systems. 25: 829-839.

[2] Abdullah A, Othman M, Sulaiman MN, Ibrahim H, Othman AT (2008). Towards a scalable Scientific Data Grid model and services. Int. Conf. Comput. Commun. Eng., pp. 20-25.

[3]International Conference on Software Engineering and Computer Systems, Pahang, Malaysia, ICSECS, Part 2 180, pp. 232-243, 2011.Springer-Verlag Berlin Heidelberg 2011.
12, 2011).

[4] Bsoul M, Khasawneh AA, Abdallah EE, Kilani Y (2010). Enhanced Fast Spread Replication strategy for Data Grid. Journal of Network and Computer Applications j.jnca.2010:12.006.

[5] Chen Y, Sun X, Wu, M (2008). Algorithm-system scalability of heterogeneous computing. J. Parallel Distr. Comput., 68: 1403-1412.

[6] Chi CH, Su M, Liu L, Wang HG (2006). An Active Peer-to-Peer System for Heterogeneous Service Provisioning. IEEE Int. Conf. Inf. Reuse Integrat., pp. 17- 22.

[7] Chidambaram J, Rao PAN, Aneesh CS, Prabhu CSR, Wankar R, Agarwal A (2008). A Methodology for High Availability of Data for Business Continuity Planning / Disaster Recovery in a Grid using Replication in a Distributed Database. TENCON, IEEE Region 10 Conference, pp.1-6.

[8]Beg AH, A. Noraziah , Abdalla AN, Mohd ZN, Sultan EI (2011). Synchronous Replication: Novel Strategy of Software Persistence Layer Supporting Heterogeneous System.

[9] Dewald B, Kline K (2011). SQL Server: Transaction and Locking Architecture. Retrieved from http://www.informit.com/articles/article.aspx?p=26657.

[10] Elghirani A, Zomaya AY, Subrata R (2007). An Intelligent Replication Framework for Data Grids. EEE/ACS Int. Conf. Comput. Syst. Appl., pp. 351- 358.

[11] Gutzait M (2007). Simplify SQL Server replication. Retrieved from http://searchsqlserver.techtarget.com/tip/Simplify-SQL-Server-replication (June 23, 2010).

[12] Guest editorial (2005). Heterogeneous computing. Parallel Computing, 31: 649-652.

[13] Gu X, Lin W, Veeravalli B (2006). Practically Realizable Efficient Data Allocation and Replication Strategies for Distributed Databases with Buffer Constraints. IEEE trans. parallel distr. syst., 17(9): 1001-1013. Beg et al. 5975

[14] Ibison P (2010). Basic Comparison of Replication Times between Merge and Transactional Replication. Retrieved from http://www.replicationanswers.com/ReplicationTimesArticle.asp.

[15] Jose M, Perez, Felix GC, Carretero J, Alejandro C, Fernandez J (2010). Branch replication scheme: A new model for data replication in large scale data grids. Future Gene. Comput. Syst., 26: 12-20.

[16] Lei M, Vrbsky SV, Hong X (2008). An on-line replication strategy to increase availability in Data Grids. Future Gene. Comput. Syst., 24: 85- 98.

[17] Li Z, Shen H (2009). A mobility and congestion resilient data management system for distributed mobile networks. 6th IEEE Int. Conf. Mobile Adhoc Sensor Syst. Macau., pp. 60-69.

[18] Lou YS, Wang ZJ, Huang L, Yue L (2009).The Study of a Reflected Persistence Data Layer Framework.WRI World Congress Software Eng., pp. 291-294.

[19] Ma D, Zhang W, Li Q (2004). Dynamic Scheduling Algorithm for Parallel Real-time Jobs in Heterogeneous System. The Fourth Int. Conf. Comput. Inf. Technol., pp. 462- 466.

[20] Noraziah A, Deris MM, Saman MYM, Norhayati R, Rabiei M, Shuhadah WNW (2009). Managing Transaction on Grid-Neighbour Replication in Distributed System. Int. J. Comput. Math. Taylor Francis., 86(9): 1624-1633.

[21] Osrael J, Froihofer L, Chlaupek N, Goeschka KM (2007). Availability and Performance of the Adaptive Voting Replication Protocol. The Second Int. Conf. Avail. Reliab. Secur. Vienna. pp. 53-60.

[22] Poddar S (2003). SQL server transactions and error handling. Retrieved from http://www.codeproject.com/ KB/database/sqlservertransactions.aspx (July 24, 2010)

[24]Bolton D (2011). About.com Guide. Retrieved from http://cplus.about.com/od/glossar1/g/multithreading.htm(February

[25]Boyera WF, Hurab GS (2005). Non-evolutionary algorithm for scheduling dependent tasks in distributed heterogeneous computing environments. J. Parallel Distr. Comput., 65: 1035-1046.

An Efficient Page Replacement Algorithm

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***Abstract*— In this paper we studied different page replacement algorithms and compared their performance. We proposed a new page replacement algorithm that uses the combination of sequential and LRU methods to obtain the better performance than that of various existing methods. We also give our attention to substitute some former method characteristic based on new idea. Key concept is that the replacement algorithm should reduce the page fault of system.**

Keywords—Main memory, cache memory, page replacement algorithms , reducing page fault.

I. INTRODUCTION

In computer operating systems, paging is one of the memory-management schemes by which a computer can store and retrieve data from secondary storage for use in main memory. In the paging memory-management scheme, the operating system retrieves data from secondary storage in same-size blocks called pages. A page fault is a trap to the software raised by the hardware when a program accesses a page that is mapped in the virtual address space, but not loaded in physical memory.

Memory is an important resource that must be carefully managed. Since main memory is usually too small to accommodate all the data and programs permanently, the computer system must provide secondary storage to back up main memory. Memory consists of a large array of words or bytes, each with its own address. The CPU fetches instructions from memory according to the value of the program counter. These instructions may cause additional loading from and storing to specific memory addresses.

A **CPU cache** is a memory used by the central processing unit of a computer to reduce the average time to access memory. Cache memories [1] [2] [3] are used in modern, medium and high-speed CPU to hold temporarily those pages of main memory .The cache is a smaller, faster memory which stores copies of the data from the most frequently used main memory locations. As long as most memory accesses are cached in memory locations, the average latency of memory accesses will be closer to the cache latency than to the latency of main memory. When the processor needs to read from or write to a location in main memory, it first checks whether a copy of that data is in the cache. If so, the processor immediately reads from or writes to the cache, which is much faster than reading from or writing to main memory. The data cache is usually organized as a hierarchy of more cache levels L1, L2, etc. In the memory hierarchy, position of cache is between processor and main memory (RAM). If the requested page is not present in cache, this event is called a cache miss or if present in cache , this event is called a cache hit . If the requested page is not present in main memory , this event is called a main memory miss or if present in main memory , this event is called a main memory hit .

We have studied different types of algorithms of page replacement such as FIFO, Optimal, LRU, NRU, Clock, LRU etc. In this paper we work to reduce the page fault as well as to reduce the overhead to system.

II. RELATED WORK

Cache memory is important for central processing unit, used to reduce the average access time of CPU. It is positioned in between the main memory and CPU[1][2][3]. Each page replacement algorithm can reduce number of page faults and also may reduce overhead of the system. Another low-overhead page replacement algorithm is **FIFO** (**First-In, First-Out**) algorithm. FIFO page replacement algorithm associates with each page the time when that page was brought into memory.when a page must be replaced the oldest page is chosen.The most unexpected condition occurs in FIFO is Belady's anomaly[8].The politic considered in NRU, is that a page without any change, during its residence in primary memory, can be known as a desirable page to be expelled from the memory. This algorithm uses two status bits named Reference bit (R) and Modification bit (M). However it is unnecessarily inefficient, since it is constantly moving pages around on its list. But a better approach can be keeping all the page frames on a circular list, in a clock form [6] .The old page will be pointed and when a page fault occurs then the pointed page will be investigated for page replacing operation. An optimal page replacement algorithm has the lowest page fault rate for any page reference stream, it simply replace the page that will not be used for the longest period of time. LRU replacement associates with each page the time of that page's last use [8].

III. LITERATURE SURVEY

A. The FIFO Page Replacement

According to this algorithm, when a page fault occurs, if there is one or more empty frame(s) in primary memory, the new invoked page will be loaded into it or one of them. But for cases in which no empty frames exist, a page with the most memory residence time will be selected for exit. In the other word, a page that has entered to the memory before all other pages, will exit the memory before them. The idea behind this politics is that, the first entered page has had enough chance to be used and this chance must be given to another page [5] . This algorithm suffers from some disadvantages. As the first, if a page is used frequently in several time periods, it will be identified as the last or oldest
page, ultimately, and may be selected to be moved out from the memory, while there is a considerable probability for urgent need to it. In such these cases, the selection will be inefficient; since the removed page must be reloaded into memory almost immediately. Another disadvantage for this algorithm relates to this fact that increasing the memory frames designated for a process can yield to a lower page fault ratio; but Belady, Flone and Shelder found that, in using FIFO for special page invoking sequences, increasing the frames number will increase the page fault ratio. Such this event is referred to as FIFO Anomaly. Figure-1 and figure-2 depict the FIFO execution for different frames number.

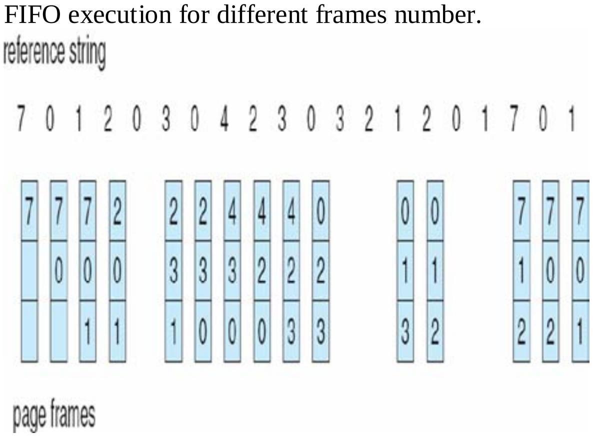

Figure- 1

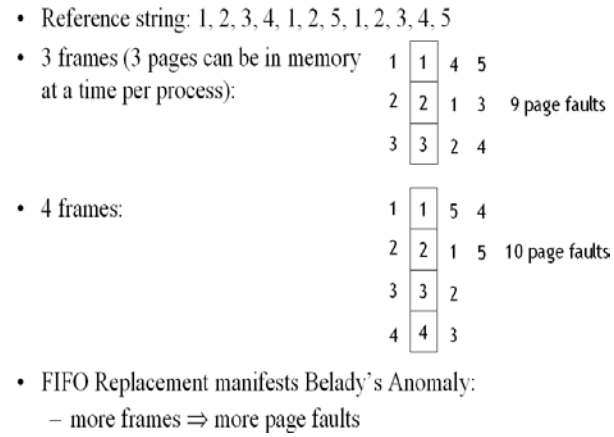

Figure – 2 : FIFO anomaly diagram

B. The NRU Page Replacement Algorithm

The politic considered in NRU, is that a page without any change, during its residence in primary memory, can be known as a desirable page to be expelled from the memory. This algorithm uses two status bits named Reference bit (R) and Modification bit (M). These bits are contained in each page table entry, and the algorithm initializes them with zero, for all pages. When a page is referred to or its contents change, R or M will be set, respectively. Since, these bits must be updated for each page referring, it is essential that they be set by the hardware. When there is a need to replace a page with a new one, first, it is attempted to find a page without any reference (R=0). If no such this page was found, a page with R=1, preferably with M=0 (without change) will be selected. The reason for such this selection is that removing pages with change (M=1), impose a secondary memory rewriting overhead. Since the reference bit for most pages is set to one, gradually, the ability to detect the most appropriate page, for exiting the primary memory, will be reduced. To preventing such this challenge, all pages reference bits are reset periodically. Regarding different states for R and M, four major groups of pages can be imagined:

Class 0: Not Referenced, Not Modified (R=0, M=0).
Class 1: Not Referenced, Modified (R=0, M=1).
Class 2: Referenced, Not Modified(R=1, M=0).
Class 3: Referenced, Modified(R=1, M=1).

The groups with lower numbers include the pages with more priority to exit the memory, and the later group's members have minimum priority. The contradictory state for class 1 is via the

C. The Clock Algorithm

The second chance algorithm could be known as a reasonable algorithm. However it is unnecessarily inefficient, since it is constantly moving pages around on its list. But a better approach can be keeping all the page frames on a circular list, in a clock form [6] . As can be seen in figure - 3, the oldest page is pointed by a hand. When a page fault occurs, the page being pointed to by the hand is investigated. If its reference bit (R) is 0, the page is moved out of memory, and is replaced with the new page. Then the hand is advanced one position. Otherwise, if R=1, it is cleared and the hand is advanced to the next page. This process is repeated until a page with R =0 is found. This algorithm, so called *clock*, differs from second chance algorithm, only in the implementation [7].

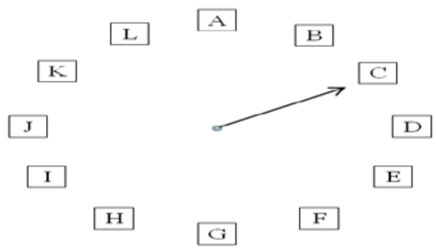

Figure – 3 : The clock algorithm schematic form

D. The Optimal Page Replacement Algorithm

An optimal page replacement algorithm has the lowest page fault rate for any page reference stream, it simply replace the page that will not be used for the longest period of time. The problem her is to know the future perfectly. Figure- 4 shows an example of this algorithm.

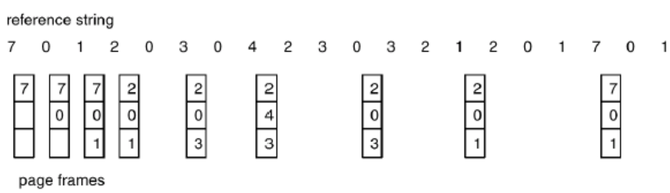

Figure - 4

E. The LRU Page Replacement Algorithm

LRU replacement associates with each page the time of that page's last use [8]. A good approximation to the optimal algorithm is based on the observation that pages that have been heavily used in the last few instructions will probably be heavily used again in the next few. Conversely, pages that have not been used for pages will probably remain unused for a long time. This idea suggests a realizable algorithm: when a page fault occurs, throw out the page that has been unused for the longest time. This strategy is called Least Recently Used (LRU) paging page replacement algorithm. Figure.5 depict the LRU execution for different frames number.

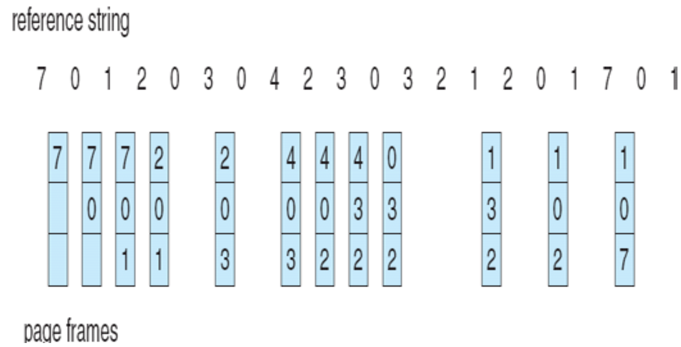

Figure - 5

IV. PROPOSED APPROACH

We propose a new algorithm for page replacement. The technique is as follows:

- If the requested page is not available in main memory and also not available in cache memory then page fault will occur .In this situation the page will be loaded from secondary storage to main memory and also keeps a copy in cache memory. In our methodology, loading of page follows the sequential manner for both of main memory and cache memory ,until they become full. Only When the cache memory is full and page fault occurs the page replacement in cache memory uses LRU algorithm.

- If the page is available in main memory (no page fault occur) the page is stored in a temporary variable first and then the page of cache memory using the LRU algorithm will send to the last location of buffer in main memory if it is not available in main memory (when the frame array is full). At the last step, the content of temporary variable will be stored in cache memory in the position from where the page number has been brought for buffering in main memory. If continuously it occur then brought page also stored in continuously at the last position until the last occur and also pointer of main memory buffer will not be set to initial in that cases. If continuously not occur after execution of above works the pointer of buffer of main memory will be reset to the initial position of the buffer. Thus the process will be continued.

A. Proposed Algorithm

1. Input reference string size , main memory and cache memory size ; // "n,f,c are reference string length , //main memory frame size and cache memory frame // respectively."
2. Input reference string in ref[] ; // " ref[] , mm[],cm[] are //arrays for reference string , main memory ,cache //memory respectively."

3. //u,v,g,s,ml,cl,pf are temporary variable
4. //ml=main memory location, cl= cl=cache memory location
5. For i=0 to n-1 do
6. Set s:= required page.
7. Find page fault for required page
8. Search page in main memory
9. Search page in cache memory

B. Description of Proposed Approach with example

Let , size of cache memory is =3

Size of main memory =6

Reference string: 1 2 3 4 5 6 7 8 1 2

From step 1 to 3 the page fault occurs and the pages stored sequentially both in main memory and cache memory.

In step 4 to 6 when the page fault occurs the page has been replaced in cache memory following LRU and stored in main memory sequentially.

In step 7 and 8 when the frame numbers are full the requested page will be loaded in last position and this process will continue till page fault occur.

In step 9 and 10 the requested page is available in main memory so there is no page fault and the page will be stored in a temporary variable first and then the page of cache memory following the LRU algorithm will be stored in the last location of frame array in main memory (when the frame array is full).At the end, content of temporary variable will be stored in the position from where the page number has been transferred from the cache memory to main memory. After execution of all steps the pointer of frame array will reset to the initial position of the memory buffer. Thus the process will be continued

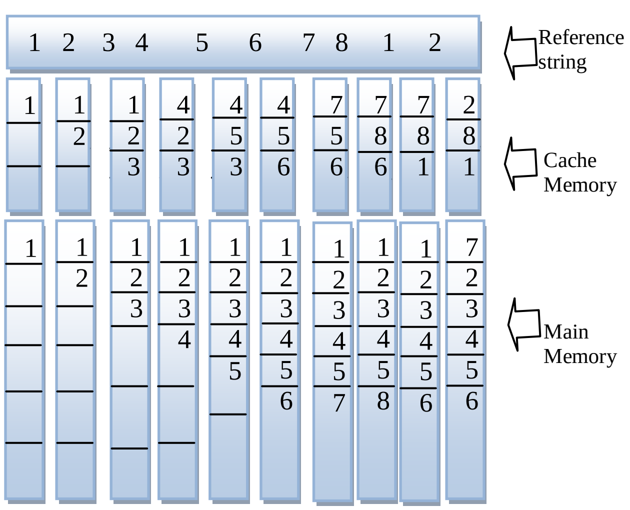

Figure: 5

3.3 Comparison between LRU and Newly Approached Algorithm

Reference String :1 2 3 4 5 6 7 8 1 2	
Frame number (Main memory)=6 Cache memory frame number=3	
LRU	**Our Approach**
Total number of page fault=10	Total number of page fault=8

Figure: 6

V. CONCLUSION

In this paper, we have proposed a new page replacement algorithm, which is essentially a modified version of the celebrated LRU algorithm. The basic distinction of our approach with the existing page replacement algorithm is that in the new one, the pages in cache and RAM always remain synchronized, whereas, in the former, they remain asynchronized. We have shown that the page fault rate in our developed algorithm is much lower than that of LRU and thus it performs better.

REFERENCES

[1] John S. Liptay. Structural aspects of the system/360 model 85 ii: The cache. IBM Systems Journal, 7(1):15-21, 1968.

[2] Gururaj S. Rao. Performance analysis of cache memories. J. ACM, 25(3):378-395, 1978.

[3] William D. Strecker. Cache memories for pdp-11 family computers. In ISCA, pages 155-158, 1976.

[4] Andrew S. Tanenbaum. Modern Operating Systems. Prentice-Hall, 1992.

[5] Hongliang, G., Chris, W., "A Dueling Segmented LRU Replacement Algorithm with Adaptive Bypassing", JWAC 2010 - 1st JILP Worshop on Computer Architecture Competitions: Cache Replacement Championship (2010), pp. 213 – 216, 2010.

[6] Ali Khosrozadeh, Sanaz Pashmforoush, Abolfazl Akbari, Maryam Bagheri, Neda Beikmahdavi "A Novel Page Replacement Algorithm Based on LRU " *J. Basic. Appl. Sci. Res.*, 2(10)10377-10383, 2012

[7] Brotherton, J., Torng, E., "Victim Caching with Better Block Replacement", Department of Computer Science, Michigan State University, pp. 162 – 182, 1996.

[8] Operating System Principles(6th edition) Abraham Silberschatz, Peter Bare Galvin, Greg Gagne
[9] Modern operating system (2nd edition) Andrew S. Tanenbaum
[10] Operating Systems: A Concept-based Approach (2E) D.M. Dhamdhere

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A Modified Approach for Selecting Optimal Initial Centroids to Enhance the Performance of K-means

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***Abstract*— Cluster analysis is one of the major data analysis tools to identify the behavior of data in a set of data items. The main purpose of this technique is to group data with maximum similarities into same clusters and separate data with dissimilarities into different clusters. K-means is one of the major clustering techniques that is widely used. The main problem of K-means is random selection of centroids. Clustering performance of the K-means completely depends upon the correctness of the initial centroids. In general, K-means randomly selects initial centroids which often show in poor clustering results. This paper has proposed a new approach to optimizing the designation of initial centroids for K-means clustering. We propose a modified approach for selecting initial centroids of K-means based on the sum score(ss) of the dataset. According to our experimental results the new approach of K-means clustering algorithm reduces the total number of iterations, improve the time complexity and also it has the higher accuracy than the standard k-means clustering algorithm.**

Keywords- Clustering, K-means algorithm,Sum Score, Data analysis, Initial centroids, Improved K-means.

I. INTRODUCTION

Clustering is the process of grouping data into a set of disjoint classes called cluster. It is an effective technique used to classify collection of data into groups of related objects. K-means clustering algorithm is one of the most widely used clustering techniques. It is a process of a set of data objects into disjoint clusters. Clustering is an example of unsupervised classification. It is similar to classification as it groups data but unlike classification groups are not predefined [2].

Clustering has been used in many application domains, including biology, medicine, anthropology, sensor networks, marketing and economics [2]. Clustering applications include plant and animal classification, disease classification, image processing, pattern processing and document retrieval. Medical taxonomy is one of the first domains in which clustering was used. In recent times classifying web log data to detect usage patterns is another important application [2].

In literature, a number of clustering methods are introduced. K-means is one of the major clustering techniques that is widely used and most popular in this category. The K-means algorithm is an effective one in computing clusters for huge practical applications in current researches such as bioinformatics, biomedical data analysis, pattern recognition etc [8,9]. Due to high computational complexity of the basic K-means algorithm, especially for large dataset, it is not time efficient and does not scale well. Moreover, it results in different types of clusters depending on the random choice of initial centroids. Researchers made a number of attempts to improve the performance of k-means algorithm. In this paper we propose a new method that improve the time complexity, cluster accuracy of k-means as well as improve the efficiency of K-means Clustering algorithm.

II. LITERATURE REVIEWS

Although K-means is a very simple and widely used clustering algorithm for variety of data types. Performance of K-means clustering algorithm strongly depends upon the selection of initial centroids. Therefore, it is quite important for K-means clustering to select initial centroids. In this paper, some proposals are reviewed.

Likas et al. [3] proposed the global k-means clustering algorithm that constructs initial centers by recursively partitioning data space into disjoints subspaces using a k-d trees method. The cutting hyper plane used in the method is defined as the plane that is perpendicular to the highest variance axis derived by principal component analysis. The partitioning is performed until each of the leaf nodes (bucket) contains less than a predefined number of data instances (bucket size) or the predefined number of buckets has been created. The centroids of data in the final buckets are then used as initial centers for K-means.

S.S. Khan and A. Ahmad [4] proposed cluster center initialization algorithm (CCIA) based on considering values for each attribute of the given data set. This can provide some information leading to a good initial cluster center.

Fang Yuan et al. [5] proposed a systematic method for finding the initial centroids. The centroids obtained by this method are consistent with the distribution of data. Hence it produced clusters with better accuracy, compared to the original k-means algorithm. However, Yuan's method does not suggest any improvement to the time complexity of the k-means algorithm.

Xu et al. [6] specify a novel initialization scheme to select initial cluster centers based on reverse nearest neighbor search.

Nazeer et al. [7] proposed an enhanced K-means algorithm, which combines a systematic method for finding initial centroids and an efficient way for assigning data points to cluster. This method ensures the entire process of clustering in $O(n^2)$ time without sacrificing the accuracy of clusters.
conference.

III. Overview of Standard K-Means Clustering Algorithm

In this section, we briefly describe the K-means algorithm. The basic idea of K-means algorithm is to classify the dataset D into k different clusters where D is the dataset of n data; k is the number of desired clusters. The algorithm runs in two basic phases[10]. The first phase is to select the initial centroids for each cluster arbitrary. In the second and final phase calculate distance of each data point with every centroid and assign data points to a cluster with nearest distance with centroids [10]. To measure the distance between data points and centroids Euclidean Distance method is used. When a new point is assigned to a cluster the cluster mean is immediately updated by calculating the average of all the points in that cluster [2]. The process of assigning a data point to a cluster and updating cluster centroids continues until the convergence criteria is met or the centroids don't differ between two consecutive iterations. Once a situation is met where centroids don't move anymore the algorithm ends. The Pseudo code for k-means clustering algorithm is given below [2].

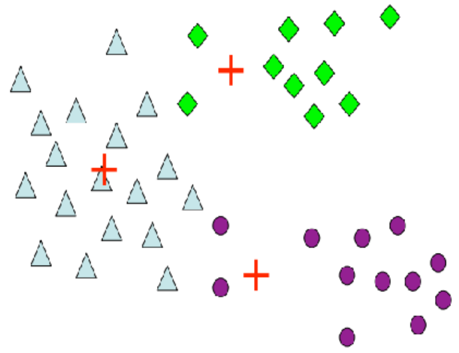

Figure 1: Random selection of initial centroids in K-means

A. *Standard K-means Algorithm*

Input:
$D = \{t_1,t_2,t_3,\ldots\ldots,t_n\}$ // set of elements
k // number of desired clusters;
Output:
K // set of clusters
Outer:
K-means algorithm:

Assign initial values for means $\mathbf{m_1},\mathbf{m_2},\ldots.,\mathbf{m_n}$;
Repeat
Assign each item $\mathbf{t_i}$ to the cluster which
has the closest Euclidean distance with mean;

Calculate new mean for each cluster.
Until
Convergence criterion is met.

IV. Proposed K-Means Algorithm

This paper proposed a novel approach to find the optimal initial centroids.To find initial centroids we calculate Sum Score (SS) of each data point.The added value of each attributes in a datapoints called sum score of a datapoint.

A. *Sum Score (SS) Calculation:*

Attributes $=x_1, x_2, x_3,\ldots\ldots\ldots\ldots\ldots\ldots,x_m$
Data Points $=T_1, T_2, T_3,\ldots\ldots\ldots\ldots\ldots\ldots..,T_n$
Sum Score (SS) $T_n = x_1+x_2+x_3+\ldots\ldots\ldots\ldots+x_n$
=

Hence, Sum Socre (SS) of
datapoint T_1 =2.50
datapoint T_2 =2.40

TABLE 1: Calculation of Sum Score (SS)

	X_1	X_2	X_3	X_4	X_5	X_6	X_7	Sum Score (SS)
T_1	0.1	0.4	0.3	0.1	0.3	0.5	0.8	2.50
T_2	0.2	0.3	0.2	0.3	0.4	0.1	0.9	2.40

A sorting algorithm is applied to sort the data points based on Sum Score (SS) and sorted data divided into k subsets where k is the number of desired clusters. Calculate mean of each subset and finally choose an initial centroids whose Sum Score (SS) is closest to the mean value.

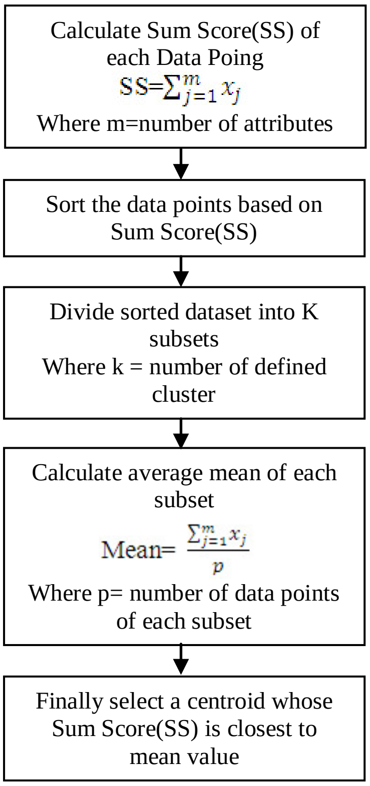

Figure 2: Flow Chart of proposed method.

For example a data set D is consists of n data points such as $T_1,T_2,T_3,\ldots\ldots,T_n$. Each data point of this set may contain multiple attributes such as T_n may contain attributes $x_1,x_2,x_3,\ldots\ldots,x_m$, where m is the number of attributes.

B. Pseudo code of Proposed Method

Input:
D = $\{t_1, t_2, t_3,\ldots\ldots,t_n\}$ // set of elements
k = number of desired clusters;
Output:
Set initial centroids K.
Steps:
1. Calculate the Sum Score(SS) of each data point;
   - $T_n=x_1,x_2,x_3,..,x_m$
   - Sum Score(SS) of $T_n = \sum_{j=1}^{m} x_j$

// where x = the attributes value, m = number of attributes, T= Data points.
2. Sort the data points based on Sum Score (SS);
3. Divide the datasets into k subsets;
4. Calculate the mean value of the each subset;
5. Select an initial centroids whose Sum Score (SS) is closest to the mean value of subsets;

The method described above to find initial centroids of the clusters is more significant than the standard k-means where centroids are selected randomly. The algorithm meets the convergence criteria faster than the standard k-means.

V. COMPLEXITY ANALYSIS

In basic K-means algorithm, the initial centroids are randomly calculated. For that, the cluster centroids are tuned many times before the convergence criterion of the algorithm is met and the data points are assigned to their nearest centroids. Since, complete reassignment of data points takes place according the new centroids, this method takes time O(nkl) where n is the number of data points, k is the number of clusters and l is the number of iterations. The proposed algorithm discussed in this paper works in two phases. In the first phase of the algorithm the time required to calculate the Sum Score (SS) of all the data points is O(n) where n is the number of data points. The algorithm then proposes to sort the data in ascending order. Sorting the data points based on the Sum Score (SS) of each data point can be done in $O(n\log_n)$ time using Merge Sort. Finally in the first phase of the proposed, the overall time complexity is $O(n\log_n)$.

The second phase of the proposed algorithm follows that of the original k-means algorithm. Distribution of the data points to the nearest cluster and the consequent tuning of centroids are conducted repeatedly until the convergence criteria reached. This process concluded with a time complexity of O(nkl) where the symbols represent the meaning mentioned above. The experimental data shows that the algorithm converges in less number of iterations as the initial centroids are calculated in a strategic way rather than randomly. Thus the overall complexity of the proposed algorithm is of $O(n(kl + \log_n))$.

VI. EXPERIMENTAL RESULTS

The multivariate data sets, taken from the UCI repository of machine learning databases that are used to test the accuracy and efficiency of the modified k-means algorithm. The same data sets are given as input to the standard k-means algorithm and the modified k-means algorithm. The value of k, the number of clusters, is taken as 3. We have applied our proposed algorithm on several datasets and compared our results with standard k-means algorithm in terms of the accuracy of cluster and total execution time.

Number of times pregnant	Plasma glucose concentration a 2 hours in an oral glucose tolerance test	Diastolic blood pressure (mm Hg)	Triceps skin fold thickness (mm)	2-Hour serum insulin (mu U/ml)	Body mass index (weight in kg/(height in m)^2)	Diabetes pedigree function	Age (years)	Class variable (0 or 1)
6	148	72	35	0	33.6	0.627	50	1
1	85	66	29	0	26.6	0.351	31	0
8	183	64	0	0	23.3	0.672	32	1
1	89	66	23	94	28.1	0.167	21	0
0	137	40	35	168	43.1	2.288	33	1
5	116	74	0	0	25.6	0.201	30	0
3	78	50	32	88	31	0.248	26	1
10	115	0	0	0	35.3	0.134	29	0
2	197	70	45	543	30.5	0.158	53	1
8	125	96	0	0	0	0.232	54	1
4	110	92	0	0	37.6	0.191	30	0
10	168	74	0	0	38	0.537	34	1
10	139	80	0	0	27.1	1.441	57	0

TABLE 2 : PIMA Indians Diabetes Data Set

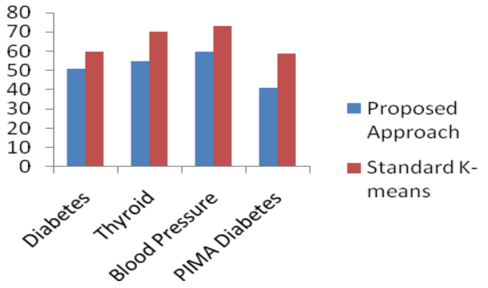

Figure 3: Iteration comparison between Proposed and Standard k-means Clustering

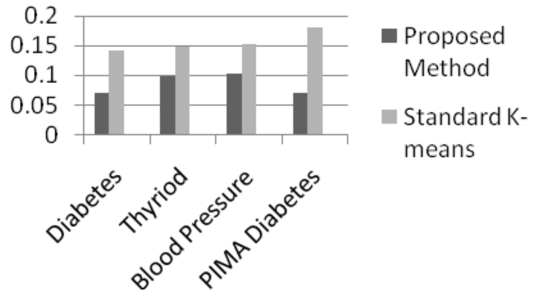

Figure 4: Time comparison between Proposed and Standard k-means Clustering Algorithm

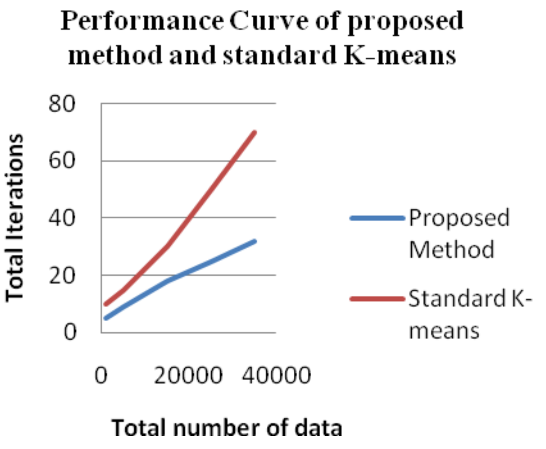

Figure 5: Iteration Comparison curve between Proposed method and standard K-means

The experimental results are shown in Table 3.

TABLE 3 : Performance analysis of proposed Algorithm

Data Sets	Number of Clusters	Total number of iterations	Algorithm: K-Means Avg. Time Taken (ms)	Algorithm: Modified Algorithm Avg. Time Taken (ms)
Diabetes	3	20	0.1423	0.0712
Thyroid	3	25	0.1493	0.0981
Blood Pressure	3	30	0.1523	0.1024
PIMA Diabetes	3	35	0.1713	0.1203

In standard k-means algorithm centroids are taken randomly but in proposed algorithm the dataset and the value of k are the only inputs needed since the initial centroids are computed automatically and find optimal cantroids by the program. In experiments proposed k-means algorithm shows better performance than standard k-means algorithm.

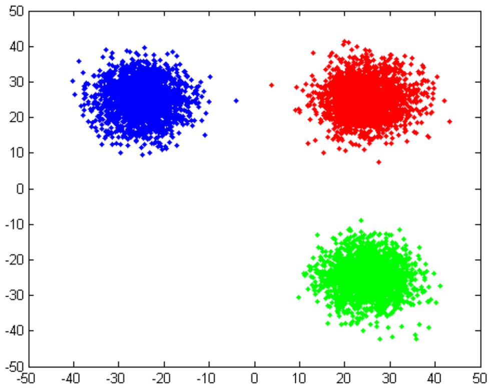

Figure 5: Optimal selection of initial centroids in proposed method

VII. CONCLUSIONS AND FUTURE WORKS

K-means algorithm is a common and widely used technique for clustering. In recent, due to incredible growth of multi-dimensional data, conventional k-means technique is inadequate to efficiently classify the distribution of data. Conventional k-means algorithm doesn't uses any technique to select initial centroids whereas accuracy of a cluster mostly depends on selection of initial centroids. So researchers nowadays are emphasized to develop new techniques to meet the raised requirements.

In this paper we proposed a new technique to select initial centroids that increase the cluster accuracy as well as decrease the time complexity also reduce the iteration time. Automating the value of k is suggested as a future work.

REFERENCES

[1] Kathiresan V., Dr. O Sumanthi, "An Efficient Clustering Algorithm based on Z-Score Ranking method", International Conference on Communication and Informatics (ICCCI-2012), Jan. 10-12, 2012, Coimbatore, INDIA.

[2] Margaret H. Dunham, S. Sridhar, "Data Mining, Introductory and Advanced Topics", Pearson Education, ISBN: 0130888923.

[3] Likas, N. Vlassis and J.J. Verbeek, "The Global k-means Clustering algorithm", Pattern Recognition , Volume 36, Issue 2, 2003, pp. 451- 461.

[4] S. S. Khan and A. Ahmad,"Cluster Center Initialization for K-mean Clustering", Pattern Recognition Letters, Volume 25, Issue 11, 2004, pp. 1293-1302.

[5] Yuan F, Meng Z. H, Zhang H. X and Dong C. R, "A New Algorithm to Get the Initial Centroids," Proc. of the 3rd International Conference on Machine Learning and Cybernetics, pages 26–29, August 2004.

[6] Xu Junling, Xu Baowen, Zhang Weifeng, Zhang Wei and Hou Jun, 2009. Stable initialization scheme for K-means clustering, Wuhan University Journal of National Sciences, Vol. 14, No. 1, pp. 24-28.

[7] Nazeer K. A. Abdul and Sebastian M.P., 2009. Improving the accuracy and efficiency of the k-means clustering algorithm, Proceedings of the World Congress on Engineering, Vol. 1, pp. 308-312.

[8] Amir Ben-Dor, Ron Shamir and Zohar Yakini, "Clustering Gene Expression Patterns," Journal of Computational Biology, 6(3/4): 281-297, 1999

[9] Daxin Jiang, Chum Tong and Aidong Zhang, "Cluster Analysis for Gene Expression Data," IEEE ransactions on Data and Knowledge Engineering, 16(11): 1370-1386, 2004.

[10] J. Han and M. Kamber, Data Mining Concepts and Techniques, Morgan Kaufmann Publishers, San Diego, 2001.

[11] Fang Yuan, Zeng-Hui Meng, Hong-Xia Zhangz and Chun-Ru Dong ,"A NEW ALGORITHM TO GET THE INITIAL CENTROIDS" , Proceedings of the Third International Conference on Machine Laming and Cybernetics, Shanghai, 26-29 August 2004

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A Sequence Alignment Algorithm and Tools for Molecular Replacement

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***Abstract*—This paper describes a new genetic alignment algorithm and software tool for sequences that can be used for determination of deletions and substitutions. Sequence alignment is one of the most active ongoing research problems in the field of computational molecular biology. Sequence alignment is important because it allows scientists to analyze protein strands (such as DNA and RNA) and determine where there are overlaps. This overlaps can show commonalities in evolution and they also allow scientists to better prepare vaccines against viruses, which are made of protein strands. The algorithm provides several solutions out of which the best one can be chosen on the basis of minimization of gaps or other considerations. The algorithm does not use similarity tables and it performs aspects of both global and local alignment. It is also compared with other sequence alignment algorithms.**

***Keywords*— Sequence Alignment; Genetic Algorithms; Computational Biology; Algorithm Complexity**

I. INTRODUCTION

Sequence alignment refers to the problem of optimally aligning sequences of symbols with or without intersecting gaps between the symbols. The objective is to maximize the number of matching symbols between the sequences and also use only minimum gap intersection, if gaps are permitted. The sequence characters in bioinformatics can be any genic (gene sequence protein sequence), structural (morphological) or behavioral features of an organism. The DNA (deoxyribonucleic acid) sequence is a string of adenine (A), guanine (G), cytosine (C) and thiamine (T) bases and there are several approaches to the solution to the alignment problem of two DNA sequences. In general, the alignment we seek is not for all the bases of the fragment but rather of various parts of it at different locations. We may find different solutions where the parts of the fragment have different gaps associated with them [1]. Given any number of sequences, finding out the degree of similarity between them is highly helpful in the field of bioinformatics, because similar regions help in deciding structural and evolutionary relationship between the sequences.

When searching in database by strings, the problem can be of errors in spellings or variant spellings or even systematic errors due to misalignment of hands with the keyboard. In such cases a similarity matrix between the characters needs to be considered. Likewise, there can be alternate spellings and representation of heard utterances in a variety of ways. In diachronic examination of and language the question of alignment must deal with changes [2]. In biological memory, fragments may be matched by means of indices [3] or by shape [4], which are aspects of alignment that go beyond string matching topics.

At present, there are various sequence alignment algorithms to find the best alignment between two sequences. In general these algorithms perform either global or local alignment or a combination of the two. Global alignment is generally performed if the sequence lengths are comparable whereas local alignment is when one is looking for the best fit within the larger sequence of the small sequence. The Needleman-Wunsch [5] and the Smith-Waterman [6] algorithms are well known algorithms for finding the best alignment between sequences. These algorithms make use of dynamic programming to find the best alignment and they use similarity tables. Needleman-Wunsch performs global optimization, whereas Smith-Waterman calculates local alignments. For sequences of length m and n, the complexity of the two algorithms in their basic forms is O(mn).

II. METHOD

A. Alignment Algorithms

The best alignment between any two given sequences easily be found by brute force if their lengths are small. If the sequence lengths are large, we must develop a strategy to minimize the comparisons.

Basically, alignment methods perform global and local alignments. We will explain global alignment with an example consisting of two sequences X and Y.

```
X: GLKATKDNCKSSEBSEFDN
   |       | | | |    |
Y: GHGFLERNCKSLMRLEDAH
```

The alignment is stretched over entire sequence lengths to match as many matches possible. Although, NCKS is the biggest match that is possible between X and Y, G and E are also considered because of the match occurrence.

Local Alignment is an alignment that searches for segments of two sequences that match really well. Local alignment stops at the end of regions of similarity. It does not take the entire sequence into consideration. It just looks for regions that have more similar segments by neglecting regions with less similar segments.

Example for Local Alignment:

X: ----------NCKS----------

| | | |

Y: ----------NCKS----------

B. *Reliable Alignment*

Examination of structural alignments shows that the more similar the proteins, the larger the proportion of the structure that can be aligned. It follows rather obviously that if one only has sequences, then the more similar the sequences, the more reliable any alignment of those sequences is likely to be. This relationship between alignment reliability and sequence similarity has been quantified.

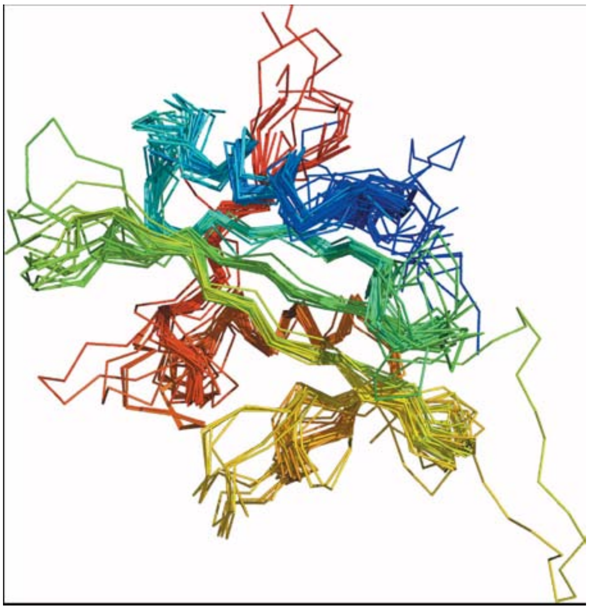

Figure 1. C_ representation of 27 SH2 Domain Structures Aligned Using the Program STAMP (Russell & Barton, 1992).

C. *Proposed Algorithm*

Our algorithm can be explained by taking two sequences into consideration. We represent our first sequence as X and second sequence as Y. Let the length of X be m and length of Y be n, In general n≤ m. In our example, we take Y to be shorter than X.

The algorithm works as follows:

Step 1: Set a=0

Step 2: Set ct1=0 and ct2=0

Step 3: If n-a ≠ 1, goto Step 4, Else goto Step 9

Step 4: Compare (n-a) size substring of Z with (m-(n-a)) substring of X

Step 5: If match found, record the positions of Y and X where match has occurred

Step 6: If (n-a) is less than n, Increment ct2, else set ct2=0 and goto Step 7

Step 7: If (m-(n-a)) is less than m, Increment ct1, else set ct1=0 and goto Step 8

Step 8: Increment 'a' by one and goto Step 3

Step 9: Print similarity regions recorded when match occurred and rest are filled with gaps(-).

D. *Example 1*

Let us consider X = CTFTALILLAGAG and Y = FTALLAAG. The length m of X is 13 and the length n of Y is 8 (where m ≥ n). Since Y is shorter, we perform comparison of sequence Y with sequence X.

These are the comparisons that occur in the first iteration of our program:

FTALLAAG is compared with CTFTALIL, TFTALILL, FTALILLA, TALILLAG, ALILLAGA and LILLAGAG.

If match does not occur, a is incremented by one, so we perform (n-a) comparison of all possible strings in Y with all possible (m-(n-a)) length strings in X.

FTALLAA and TALLAAG in sequence Y are compared with CTFTALI, TFTALIL, FTALILL, TALILLA, ALILLAG, LILLAGA and ILLAGAG in sequence X.

We perform such comparisons in sequence till (n-a) becomes 1 or all character in shorter sequence Y gets match with sequence X.

Finally, we obtain alignment as:

X: C T F T A L I L L A G A G
Z: - - F T A L - - L A – A G

The sequences as presented to us may have errors. These errors could be of different kind: substitution errors, extra characters, dropped characters. Clearly, one must correct for errors based on knowledge of the database from where the strings have come. The matching process can point to likely places of substitution, extra characters, and dropped characters.

Let us assume we have an error in sequence X. Let the actual value of X = CTFTALILLAAG obtained after error correction. Now, if we perform alignment between X and Y, we might get better alignment than what we got before. Likewise there could have been an error in the sequence Y. If errors are known to have occurred it is important to correct these errors.

In the above example, if we consider X after error correction then we get a better result than that of the alignment that we got before.

$\mathbf{X_{new}}$ = C T F T A L I L L A A G
$\mathbf{Z_{new}}$ = - - F T A L - - L A A G

Z_{new} has better alignment than Z after considering error correction.

If there exists more than one match between any two sequences, we must select an alignment that has minimal gap amongst them or use other considerations related to similarity between characters. Here we propose the following method to choose the better alignment: Generate all possible alignments between sequences and calculate the mean and variance for the gaps for each of those alignments. An alignment with less gap mean and smaller variance, if the means are the same, is the optimal alignment.

E. *Example 2*

Let us consider X= SNARSENAGCATQRABCRTLJT and Y= ARAGCATR. The possible alignments Z that we could get are given below:

X: S N A R S E N A G C A T Q R A B C R T L J T
Z: - - A R - - - A G C A T - R - - - - - - - -
- - A R - - - A G C A T - - - - - R - - - -

In the above example, we have 2 possible alignments for Z. The gap variance is computed after finding the mean of the gaps which are: 2 and 4 respectively. The variance values for the gaps are: 1 and 1 respectively. But still first one has minimum number of gaps, Therefore, the first alignment is the best one.

Some words in English dictionary, which are frequently confused with words that sound same, but are spelled differently. For example effected and affected, to and too, sent and cent, here and hear and many more. These words are called as homonyms. Same names are spelled in different ways in different regions or they are pronounced differently. The English “a” has the pronunciation of “ai” in many words. If we are able to derive a matrix that tells us which letters could be substituted with existing letters that could minimize the gaps between sequences, and our algorithm could then work significantly better.

F. *Example 3*

X=CGTCTAACTAGGTACAGTAGAG and Z=TACTAGGAG, so that m= 22 and n=9. Here are the possible alignments our algorithm generates.

X =C G T C T AA C T A G G T A C A G T A G A G
$\mathbf{Z_1}$= - - T - - - A C T A G G - - - A G - - - - -
$\mathbf{Z_2}$= - - - -T -A C T A G G - - - - - - - - - A G
$\mathbf{Z_3}$= - - - -T- A C T A G G -A - - G - - - - -
$\mathbf{Z_4}$= - - - - T- A C T A G G - - - - - - A G- -

There are many possible alignments that can be generated, Likewise, our algorithm generates all possible alignments and computes variance for all alignments. An alignment with less variance could be chosen as best alignment.

In the above example, mean of Z1, Z2, Z3 and Z4 is 4.67, 6.33, 3 and 4 respectively. Variances are 5.56, 14.89, 8.5 and 9.5. As alignment string Z1 has minimum variance, Z1 is considered as best alignment amongst Z1, Z2 Z3 and Z4. This strategy defines a new way of selecting optimal alignment between given any number of sequences. When we perform alignment between any sequences, our main objective is to minimize gaps between them. So we have to select optimal alignments by discarding all non-optimal alignments that would create more gaps. When given sequences are long, we need to use dynamic programming to calculate best possible alignment. But in the beginning, it is also necessary to prune bad alignments.

G. *Flow Chart of Software Tool*

The software tool that is developed by Java language by using the algorithm, works by following this process flow chart.

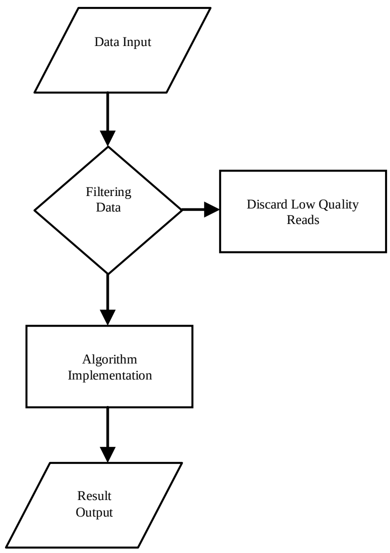

Figure 2. Flow Chart of the Process of Developed Software Tool.

III. COMPLEXITY ANALYSIS

For sequences X and Y of length m and n respectively, the proposed algorithm performs following number of comparisons to calculate alignment between them where k is an integer that ranges from 1 to n, so the complexity of our algorithm can be derived by the derivation of following,

$$\sum_{k=0}^{n} (m-(n-k)) * (n-k)$$

=(m-n)*n+((m-n-1)*(n-1))+((m-n-2)*(n-2))+-----+mn

=mn+(n*n)+mn-m-(n*n)-n-n+1+-------------------+mn

As the higher order terms are m and n, after expansion we get complexity of O(mn).

IV. COMPARISON WITH EXISTING ALGORITHM

If we consider two sequences X and Y, where X = GGCCTAACTAGAATACCGTACAGTACGAAG and Y = CACTAGGAA.

Alignment using Needleman Wunsch Algorithm:

```
GGCCTAACTAGAATACCGTACAGTACGAAG
    | |    | | | |   |                          |
    CA -CTAGGA -  - -  - - - -  - - - -  - - - - - -A
```

Alignment using Smith Waterman Algorithm:

```
GGCCTAACTAGAATACCGTACAGTACGAAG
                    | |    | | | | |
-  - -  - CA - CTAGGAA  - -  - - -  - -  - - -  - - -  - - -
```

Alignment generated by our algorithm:

```
GGCCTAACTAGAATACCGTACAGTACGAAG
            |   | | | | |           |   |   | |
        - - - - C - ACTAG - - - - - - G - A - A
```

The developed software tool gives the output as the following:

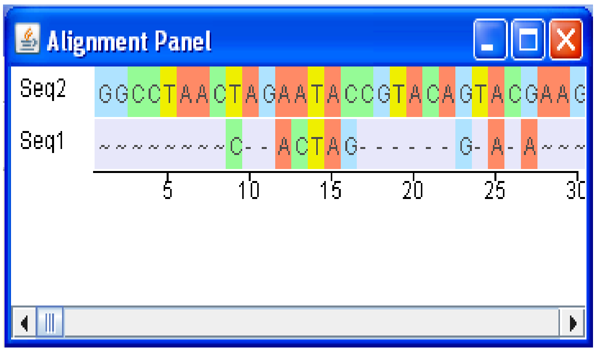

Figure 3. The output GUI of software tool developed in Java.

V. CONCLUSION

In conclusion, the goal of this paper is to describe a new alignment algorithm and software tool developed in Java for sequences that can be used for determination of deletions and substitutions. Insertions may also be handled if the order of the comparison sequences is switched but it would make sense only if the size of the two sequences is comparable. Our algorithm provides several solutions out of which the best one can be chosen on the basis of minimization of gaps or other considerations. Statistical consideration related to alignment solutions can be studied in a manner similar to those for other alignment algorithms. The algorithm and the application generates good alignment by finding maximum length matches between given sequences. Its complexity is of the same order as the Needleman Wunsch and Smith Waterman algorithms. The basic algorithm can be refined by adding further constraints related to character similarity or gap constraint that will also make it more efficient.

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REFERENCES

[1] D. Gusfield, "Algorithms on strings, trees, and sequences", *computer science and computational Biology*, Cambridge University Press, 1997.

[2] S. Kak, "A frequency analysis of the Indus script", Cryptologia 12:129-143, 1988.

[3] S. Kak, "The three languages of the brain: quantum, reorganizational, and associative", In: K. Pribram and J. King (editors), Learning as Self-Organization. Lawrence Erlbaum, Mahwah, 185 –219, 1996.

[4] L.S. Swan and L.J. Goldberg, "Biosymbols: symbols in life and mind", Biosemiotics 3: 17-31, 2010.

[5] S.B. Needleman and C.D. Wunsch, "A general method applicable to the search for similarities in the amino acid sequence of two proteins", *Journal of Molecular Biology* 48: 443–53, 1970.

[6] T.F. Smith and M.S. Waterman, "Identification of common molecular subsequences", *Journal of Molecular Biology* 147: 195-197, 1981.

[7] D. Garg and N. Singla, "String Matching Algorithms and their Applicability in various Applications", *International Journal of Soft Computing and Engineering*, Vol-1, Issue- 6, January 2012.

[8] Ziad A. A., Alqadil M. Aqel, and El Emary I. M. M., "Multiple Skip Multiple Pattern Matching Algorithm (MSMPMA)", *IAENG International Journal of Computer Science*, September, 2007.

Performance Evaluation of Warshall Algorithm and Dynamic Programming for Markov Chain in Local Sequence Alignment

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***Abstract*— Markov Chain is very effective in prediction basically in long data set. In DNA sequencing it is always very important to find the existence of certain nucleotides based on the previous history of the data set. We imposed the Chapman Kolmogorove equation to accomplish the task of Markov Chain. Chapman Kolmogorove equation is the key to help the address the proper places of the DNA chain and this is very powerful tools in mathematics as well as in any other prediction based research. It incorporates the score of DNA sequences calculated by various techniques. Our research utilize the fundamentals of Warshall Algorithm (WA) and Dynamic Programming (DP) to measures the score of DNA segments. The outcomes of the experiment are that Warshall Algorithm is good for small DNA sequences on the other hand Dynamic Programming are good for long DNA sequences. On the top of above findings, it is very important to measure the risk factors of local sequencing during the matching of local sequence alignments whatever the length.**

Keywords: Keywords:Hidden Markov Model, Chapman-Kolmogorov formula, Warshall Algorithm, Dynamic Programming, Score measurement.

I. INTRODUCTION

Markov Chain can be easily formulated in the state space for the simple model such as 0 for first Nucleotide Adenine (A), 1 for second Nucleotide Cytosine (C), 3 for Thiemann (T) and finally 4 for Guanine (G). For same data set it can be also possible by using second order Markov Chain value as {00,01,02,03,04...............}. Since the data sets in DNA sequences contain four fundamental bases, there should be 4^2 possible space states. But the complexity for higher order model is higher than a simple model and for this reason Markov process always holds the simple state space. For example if we ask to predict for 10000th Nucleotide in a sequence and we have to measure the 9999th Nucleotide in the same sequences.

On the same time it is possible to quantify the pairwise evolutionary distances, Hamming distance. If U is the total number of mismatches in an alignment of length l, then the Hamming distance for per 10000 sites is

$$H(U,l)=10000\frac{\vartheta}{l} \text{ ....(1)}$$

The equation 1 above works good when the DNA sequences space is discrete. To measure the discrete Markov Process for the A,G,C, and T Nucleotides the starting distribution will be as follows: P^0=ρC,ρA,ρG,ρT . The figure 1 below shows the basic Markov Process Transaction for Discrete system in DNA sequencing.

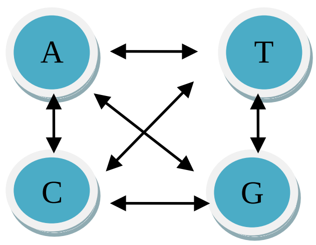

Figure 1: The transactional states for Markov chain.

II. MOTIVATION

According to the various algorithms, implemented through 2001 to 2005 as Ning [1], Kent [3,4] Schwartz [5] and Watanabe [6] examined the accuracy of the sequencing. In the age of information superhighway we know that the genome sequences may have continual or near about continual patterns in the given or collected data sets. As a result the outcomes might be same for many positions. On the contrary the mutations and **Indel** may generate incorrect judgments for sequencing and mapping whether it is local, global or pair-wise alignments or mapping. The algorithms above do not consider the situations regarding the repetitions of the patterns, mutations and incorrect mapping. Here we have noticed the system rejects the data sets, made the area small and as consequences the calculations become complicated as well as wrong. According to the Ewing and Green [2] proposed a solution to overcome the ambiguity but this method is candidate of low-quality regions. Here we have implemented the Markov chain concept under Chapman – Kolmogorove equations to established probable sequenced positions. In the field of sequencing, there are lot of software's helping to detect the sequences and the frequently used systems are PolyPhred [7], SNP detector [8], and novoSNP [9]. But these three systems only detect genotype sample. They are unable to solve the dynamic patterns and sequences. In this research we have imposed the idea on prediction using Markov Process under the support of Dynamic Programming and Warshall Graph Algorithm. The fundamental step of prediction is that this Chapman Kolmogorove calculative prediction. We have compared DP and WA algorithm in the light of Local Sequence alignment with various length. The detection based depends on homolog finding. The homolog finding mainly depends on the database finding [10]. But the database finding is not always efficient due to the size and cost of the equipments. Training based sequencing is not able to identify the proper coding regions due to the lack of generalizations [11]. Besides, predictions are vulnerable with many false positives identifications and sensitivities [14].

III. DYNAMIC PROGRAMMING

Dynamic Programming decomposes the sequences into several parts and solved the problem recursively until it reaches to a particular condition. Sometimes decompositions becomes difficult and it hard to get a clear-cut solution. In that case we should investigate several possibilities and Recursive solution is one of the acceptable methods to overcome the problem. Basically, Dynamic Programming to choose the Maximum Matching Sub Sequences (MMSS).

Suppose

$P = p_l, p_2 \ldots p_n$, and $Q = q_1, q_2, \ldots q_m$, are two DNA sequences. The indel of two sequences is denoted by the weight M (g). At first we consider the best alignments as F (p, q) = max $\forall$ (p*,q*). Where, F is a function which relates the current alignments and new alignment. By using Dynamic Programming we can check the sequences F (p,q) recursively.

$F(i,j) = F(p_l, p_2 \ldots p_i\, q_1, q_2, \ldots q_j)$

Where $F(0, O)=O$, $F(O,j)=F(-,q_1, q_2, \ldots q_j) = M(j)$ and $F(i, 0)= M(i)$. Then:

$F(i,j)=\{F(i-1,j-l)+F(p_i,q_j)$, max $\{ F(i-k, j) + M(k)\}$, max $\{ F(i, j-I) + M(l)\}$. For local sequence alignment the function $L(p, q)=\max\{F(p_u,p_{u+1},\ldots p_v,q_x,q_{x+1},\ldots q_y): 1 \le u \le v \le n, 1 \le x \le y \le m\}$.

IV. WARSHALL ALGORITHM ALIGNMENT

Warshall graph algorithm is a sequence path finding process which subgroups the entire data set into set of intermediate nodes along the path. To perform the decompositions, it is easy to label the nodes set form 1 to n. The decompositions helps to reduces the shortest paths along the sequences. By designing the total path under the variables I,J, and K we can say if there is path from I to J and J to K than we can say that there is a path from I to K. In a word we can say that the total path is

P[I,J,K]=Shortest path from I to J using the only intermediate node 1……………………………K. the recursive process of solving the shortest path is as follows:

$$P[I,J,K] = MIN\{P[I,J,K-1], P[I,K,K-1] + P[K,J,K-1]..(1)$$

We can illustrate equation 1 as follows as algorithmic steps.

Sequencing Graph (Adjacent Matrix:ADM_R, n×n,)

1. Initialize Graph weight for all nodes as Weight: $=ADM_R$,($Weight=w_{ij}$)
2. Loop k=1;to n,
3. Inner loop J=1 to n;
4. Inner loop I=1;to n,
5. $W_{ij}=w_{ij} \vee (w_{ik} \wedge w_{kj})$
6. $W_{ij}=w_{ij} \vee w_{kj})$
7. Return initial Weight

V.MARKOV CHAIN ALIGNMENT

Our motivation on mismatches identification according to the Chapman-Kolmogorove formula on Markov chain based stochastic matrix. The formula for a stochastic process with random variable X is $X=\{X_t, t \in T\}$. Where t= index and it indicate the time. X_t = State of the process .T= Index set constitute by time t.

Suppose n=0, 1, 2, 3….. And m=1, 2, 3…and $i_0 \ldots\ldots\ldots i_m \in E$. E=All possible values that the random variable X_t can assumes. Then

$$\Pr\{X_{n+1} = i_1, \ldots, X_{n+m} = i_m \mid X_n = i_0\} =$$

$$\Pr_{i_0 i_1} \cdot \Pr_{i_1 i_2} \cdot \ldots \cdot \Pr_{i_{m-1} i_m} .$$

$$\Pr\{X_{n+m} = j \mid X_n = i\}$$

$$= \sum_{k=0}^{\infty} \Pr\{X_{n+m} = j \mid X_{n+1} = k, X_n = i\} \Pr\{X_{n+1} = k \mid X_n = i\}$$

$$= \sum_{k=0}^{\infty} \Pr\{X_{n+m} = j \mid X_{n+1} = k\} \Pr\{X_{n+1} = k \mid X_n = i\}$$

$$= \sum_{k=0}^{\infty} \Pr_{ik} \Pr_{kj}$$
$$= Pr_{ij}^{m}$$

In general,

$$\Pr_{ij}^{n+m} = \sum_{k \in E} \Pr_{ik}^{n} \Pr_{kj}^{m} \quad \text{for all } n, m \geq 0, \text{ all } i, j \in E.$$

VI. IMPLEMENTATION

We have implemented and experimented under the environments of Java with Integrated Development Environment (IDE) Netbeans. The object oriented implementation helped us to perform the nucleotides (A, C, T, and G) as a distinct object. In our previous work [1] we have improved the performance of [16] and noticed our RSAM algorithm is significantly better in the light of Speed, Complexity, Space, Sensitivity, Accuracy and risk. In this research we have compared all the above parameters under the light of Markov Process and Warshall Graph Algorithm. For Speed, Sensitivity and Accuracy we have measured referential value as best, average and low. Here we have checked the complexity, risk, accuracy and space for the first time and many local sequence alignment tools measured the sensitivity without any standard parameter. According to the MUMmer [17] termed the parameter '**q**' as the ratio between accurate aligned nucleotides pairs and total number of nucleotides in the given sequence. Total Column Score (TCS) is another aspect of MUMmer procedure. Again according to the AVID [18], where the authors considered the alignment pairs which have the score greater than the predefined threshold value. Instead of all of the methods above, we have concentrated towards the set operations under the complete machine learning process on exons and introns. Introns measurement are also essential part of the alignment to maintain proper checking instead of only one parameters checking (exons). For speed, sensitivity and accuracy the reference values have been checked according to the fuzzy manner, such as: best (H), average (M) and low (L).but according to the [16] there is no clue to compare the Sensitivity of the sequencing.

VII. RESULT

The outcomes of these two process, a few interesting changes have had observed. Chapman-Kolmogorov equation in Markov process is very efficient for any arbitrary predictions in any DNA segments or sequences. It is clearly noticed that Markov Process has potential and strong capabilities to handle the data set whatever the environment. In this case we also found that MP has significanlty better scpes for long data set.

Figure 2 below shows the experimental outcomes for Markov Process.

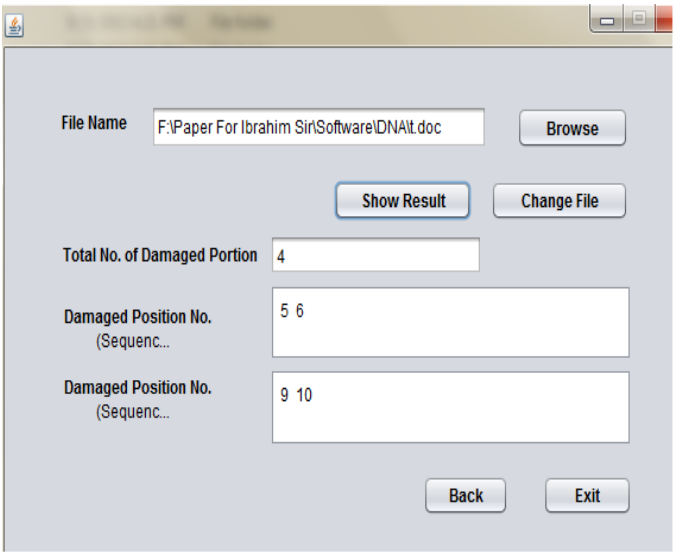

Figure 2: Markov Process out comes for data set.

On the contrary, Warshall Graph Algorithm, has limited scopes than that of Markov Process. But it has faster capabilites on short data set. The speed of the processing is sharply better than Markov Proceess. Other parameters such as complexity , Risk, Sensibity, Space and Accuracy are also signicicantly better than Markov Process due to its faster solving capabilties. Only pivotal drawback is that Warsehall Graph Algorithm work only for short length data set what ever the protein, DNA or RNA. Figure 3 below shows the performace Dynamic programming impact for Markov Process.

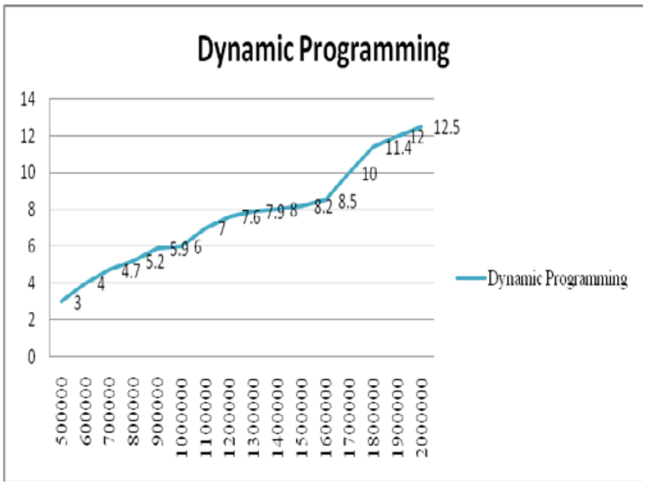

Figure 3: The impact of Dynamic Programming for Markov Process

Here the CPU Utilization time for long data set range from 1500000 to 2000000 requires are 8.2, 8.5, 10, 11.4,12, and 12.5. On the contrary, Warshall Graph algorithm takes more time for the same data set and the time values are 8.2, 9, 10.8, 12, 12.9 and 14. But for the previous data set whose lengths are less than 1500000, Warshall Graph Algorithm takes less time than Dynamic Programming. Figure 4 below shows the impact for Warshall Graph Algorithm.

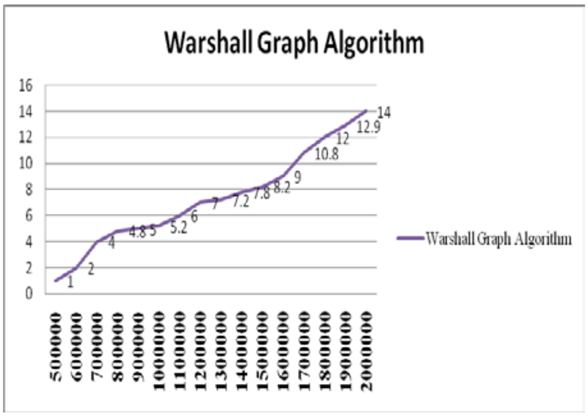

Figure 4: Impact of Warshall Graph algorithm

The reasons behind Dynamic programming requires more time to solve small data set is that it works for arbitrary probabilistic values where Warshall Graph Algorithm works deterministic path and values. The comparative results of these two methods are below at figure 5.

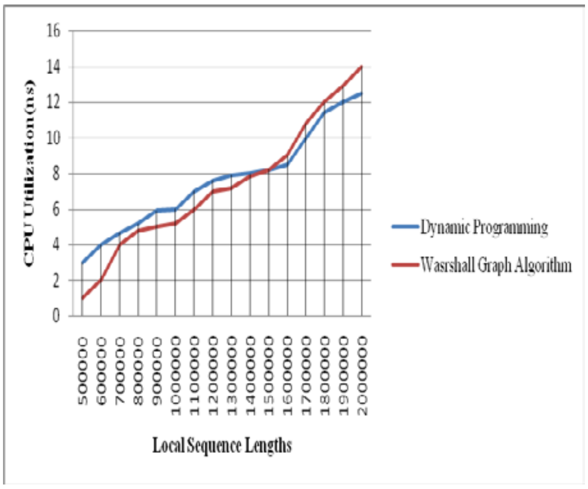

Figure 5:Comparitive Illustration of Dynamic Programming and Warshall Graph Algorithm.

VIII. CONCLUSION

Both Dynamic Programming and Warshall Graph Algorithm perform predictions of DNA base pair according to the process. Dynamic Programming has better capabilities to handle large data set due to its randomness. On the other side, Warshall Graph Algorithm works based on predefine values and path. That why Warshall Graph Algorithm has to check the entire path and values weather the path is short or long. That is the reason Dynamic Programming requires more time. We will find why Randomness causes more time and deterministic process is better for small data set in future work.

REFERENCES

[1]. Mohammad Ibrahim Khan and Md.Sarwar Kamal, ***"RSAM: AN INTEGRATED ALGORITHM FOR LOCAL SEQUENCE ALIGNMENT",*** Archives Des Sciences, Vol 66, No. 5;May 2013, ISSN 1661-464X,pp,395-412.

[2]. Ning Z., Cox A.J., Mullikin J.C. SSAHA: "A fast search method for large DNA databases", Genome Res.; 11:1725–1729, 2001.

[3]. Ewing B., Green P. "Base-calling of automated sequencer traces using *phred*. II. Error probabilities", Genome Res.;8:186–194,1998.

[4]. Kent W. J., "BLAT—the BLAST-Like Alignment Tool", Genome Res.**12**:656–664, 2002.

[5].Kent W.J., Sugnet C., Furey T., Roskin K., Pringle T., Zahler A. , Haussler D. ,"The human genome browser at UCSC", Genome Res.**12**:996–1006,2002.

[6]. Schwarz D.S., Hutvagner G., Du T., Xu Z., Aronin N., and Zamore P.D., "Asymmetry in the assembly of the RNAi enzyme complex", Cell **115:** 199,2003.

[7]. Watanabe T., Takeda A., Mise K., Okuno T., Suzuki T., Minami N., and Imai H., "Stage-specific expression of microRNAs during Xenopus development", FEBS Lett. **579:**318,2005.

[8]. Stephens M., Sloan J.S., Robertson P.D., Scheet P., Nickerson D.A., "Automating sequence-based detection and genotyping of SNPs from diploid samples", Nat. Genet.vol 38,pp.375-381, 2006.

[9]. Zhang J., Wheeler D.A., Yakub I., Wei S., Sood R., Rowe W., Liu P.P., Gibbs R.A., "Buetow K.H. SNPdetector: A software tool for sensitive and accurate SNP detection". PLoS Comput. Biol., 2005.

[10]. Weckx S., Del-Favero J., Rademakers R., Claes L., Cruts M., De Jonghe P., Van Broeckhoven C., De Rijk P. "novoSNP, a novel computational tool for sequence variation discovery", Genome Res.;15:436–442,2005.

[11]. Stephens M., Sloan J.S., Robertson P.D., Scheet P., Nickerson D.A., "Automating sequence-based detection and genotyping of SNPs from diploid samples", Nat. Genet.vol 38 (2006),pp.375-381.

[12]. Zhang J., Wheeler D.A., Yakub I., Wei S., Sood R., Rowe W., Liu P.P., Gibbs R.A., "Buetow K.H. SNPdetector: A software tool for sensitive and accurate SNP detection". PLoS Comput. Biol., 2005.

[13]. Weckx S., Del-Favero J., Rademakers R., Claes L., Cruts M., De Jonghe P., Van Broeckhoven C., De Rijk P. "novoSNP, a novel computational tool for sequence variation discovery", Genome Res.;15 (2005):436–442.

[14]. Yetisgen-Yildiz, M. & Pratt, W., "The effect of feature representation on Medline document classification", In AMIA Annual Symposium Proceedings, , American Medical Informatics Association, vol. 23 (2005), 849.

[16]H.Waqaar, A. Alex, and R. Bharath, An Efficient Algorithm for Local Sequence Alignment, 20-24, 2008**.**

[17] Delcher A.L., Kasif S., Fleischmann R.D., Peterson J., White O., and Salzberg S.L. 1999. Alignment of whole genomes. Nucleic Acids Res. 27: 2369-2376.

[18] Bray Nick, Dubchak Inna and Pachter Lior, Avid: A global alignment program, Genome Research. 2003 13: 97-102; 2002.

A New Approach of Disk Scheduling Algorithm

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Abstract --***The operating system is responsible for using resources efficiently. The disk is of course, one of the computer resources. Computer processing speed depends on disk speed. Processor speed and memory capacity is increasing several times faster than disk speed. This disparity suggests that disk I/O performance will become an important bottleneck. Disk performance management is an increasingly important aspect of operating system research and development. Scheduling is a fundamental operating system function. People have improved I/O performance by intelligent scheduling of disk. In this paper we introduce a new disk scheduling algorithm that reduce the number of head movement by proper scheduling therefore it maximizes throughput for modern storage devices.***

1. INTRODUCTION

The performance of disk I/O schedulers is affected by many factors such as workloads, file systems, and disk systems. The Disk is said to be of two basic types:

1. Fixed head disk- This has one head for each track on the disk and it requires no head movement time to service a request. This is quite expensive.
2. Movable head disk- This is much more common in use because it has a single head driven by a stepper motor that can position the head over any desired track on the disk surface.

The task of scheduling usage of sharable resources by the various processes is one of the important jobs of operating system as it is responsible for efficient use of the disk drives. The efficiency of disk drivers means that disks must have fast access time and reasonable bandwidth. The two major components of access time and bandwidth of disks are:

- *Seek time*- the time to move the heads to the cylinder containing the desired sector.
- *Rotational latency*-the additional time to rotate the desired sector to the disk head.

We can improve both the access time and bandwidth by scheduling the servicing of disk I/O requests in a good order. Whenever a process needs I/O to or from the disk, it issues a system call to the operating system. The request specifies several pieces of information:

- Whether this operation is input or output.
- Whether the disk address for the transfer is
- What the memory address for the transfer is
- What the number of sectors to be transferred is

If the disk driver and controller are available, the request can be serviced immediately. If the driver or controller is busy, any new, request for service will be placed in the queue of pending requests for the drive. For a multiprogramming system with many processes, the disk queue may often have several pending requests. Thus, when one request is complete, the operating system chooses which pending request to service next. It is now clear that there can be number of programs in memory at the same time that results in overlapping of CPU and I/O. There are batch programs that run without interaction from user. There may be time shared programs that run with user interaction. For both of these the common name used is Process for which burst cycle of CPU characterizes execution of their process, alternatively between CPU and I/O activity. The scheduling makes selection among the processes in memory that are ready to be executed and makes allocation of the CPU to one of them. The decision regarding scheduling takes place when a process switches from:

- Running to waiting state
- Running to ready state
- Waiting to ready state
- Terminates

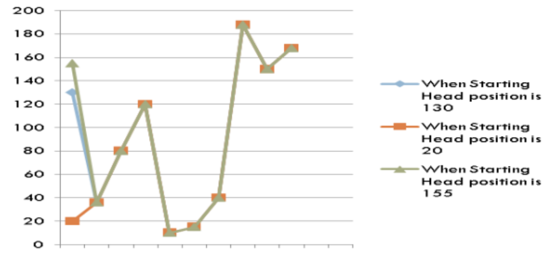
The scheduling of the above processes is known as non-preemptive. This can be considered very important in case of systems with multi programming as they have a common file system. The file system is said to be common in multi programmed systems because it is shared by all the users even though each of them may have one's own file. This common file system may be spread out over a finite number of disks or it may reside entirely on a single disk. Thus, all processes that do disk IO are competing for access to the same physical disk or set of physical disks. Mostly as any given disk can only perform one access at a particular time, if several accesses are requested on a given disk, some order of service for the requests is established by the operating system.

2. PROBLEM STATEMENT AND MOTIVATION

Several algorithms exist to schedule the servicing of disk I/O requests:

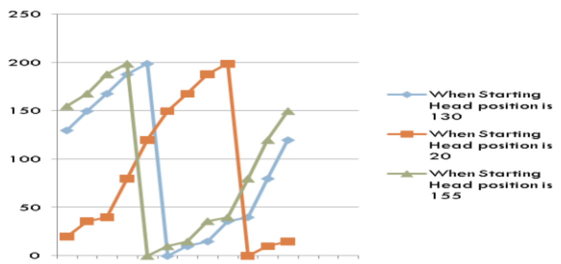
2.1 FCFS

The simplest form of disk scheduling is, of course, the first-come, first-served algorithm .All incoming requests are placed at the end of the queue. Whatever number that is next in the queue will be the next number served. Using this algorithm doesn't provide the best results. To determine the number of head movements you would simply find the number of tracks it took to move from one request to the next.
Consider, for example a disk queue with requests for I/O to blocks on cylinders: Queue (0-199): 36, 80,120, 10, 15, 40,188,150 and 168.For this case it went from 36 to 80 to 120 and so on. From 36 to 180 it moved 45 tracks. If you tally up the total number of tracks you will find how many tracks it had to go through before finishing the entire request. This schedule is diagrammed in figure 1.1



2.2. SSTF:

Shortest Seek Time First-Selects the request with the minimum seek time from the current head position. Since seek time increases with the number of cylinders traversed by the head, SSTF chooses the pending request closest to the current head position. SSTF scheduling is a form of SJF scheduling: may cause starvation of some requests.
Consider, for example a disk queue with requests for I/O to blocks on cylinders: **Queue (0-199):** 36, 80,120,10,15,40, 188,150 and 168. This schedule is diagrammed in figure 1.2



2.3. SCAN

The disk arm starts at one end of the disk and moves toward the other end, servicing requests until will get to the other end of the disk, where the head movement is reversed and the servicing continues. Some time called the elevator algorithm. . Once again this is more optimal than the previous algorithm, but it is not the best.

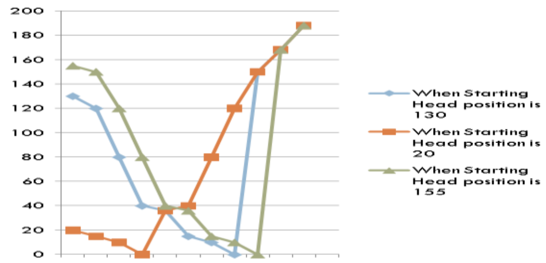
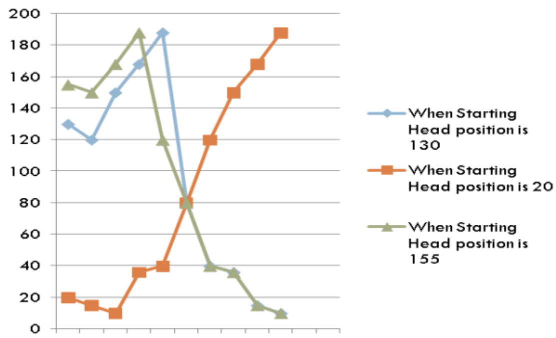


Fig2.3: Head movement of SCAN

Consider, for example a disk queue with requests for I/O to blocks on cylinders: Queue (10-199):36, 80,120,10,15,40,
188,150,168. This schedule is diagrammed in figure 1.3

2.4. C-SCAN

Circular scanning works just like the elevator to some extent. Provides a more uniform wait time than SCAN. The head moves from one end of the disk to the other. Servicing requests as it goes. However, when it reaches of the other end, it immediately will return to the beginning of the disk, without servicing any requests on the return trip.Treats the cylinders as a wrap around circular list from the first cylinder to the last one.
Consider, for example a disk queue with requests for I/O to blocks on cylinders: Queue (0-199):36, 80, 120, 10, 15, 40, 188,150, 168.This schedule is diagrammed in figure 1.4



5.5. C-LOOK

This is just an enhanced version of C-SCAN. Arm goes only as far as the last request in each direction, the reverses direction immediately, without first going all the way to the end of the disk.
Consider, for example a disk queue with requests for I/O to blocks on cylinders: Queue (0-199): 36,80,120,10,15, 40,188,150, and 168. This schedule is diagrammed in figure 1.5

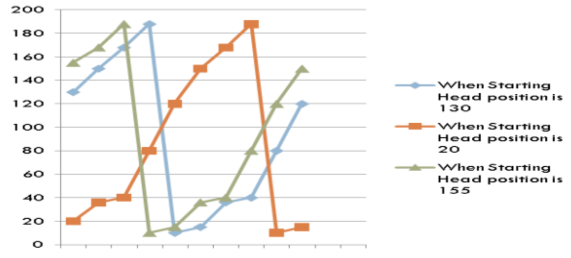


Fig 2.5: Head movement of C-LOOK

2.6. LOOK

LOOK is similar to SCAN in that the heads sweep across the disk surface in both directions performing reads and writes. However, unlike SCAN, which visits the innermost and outermost cylinders each sweep, LOOK will change directions when it has reached the last request in the current direction. Consider, for example a disk queue with requests for I/O to blocks on cylinders: Queue (0- 199): 36,80,120,10,15, 40,188,150, and 168. This schedule is diagrammed in figure 1.6

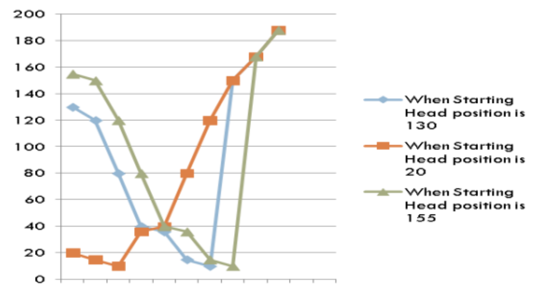


Fig:2.6 Head movement of LOOK

3. PROPOSED DISK SCHECULING ALGORITHM

Proposed Algorithm basic Theory: At first sorting in ascending order of all cylinders input blocks by using any sorting method. Find the value of path1 and path2. From the figure

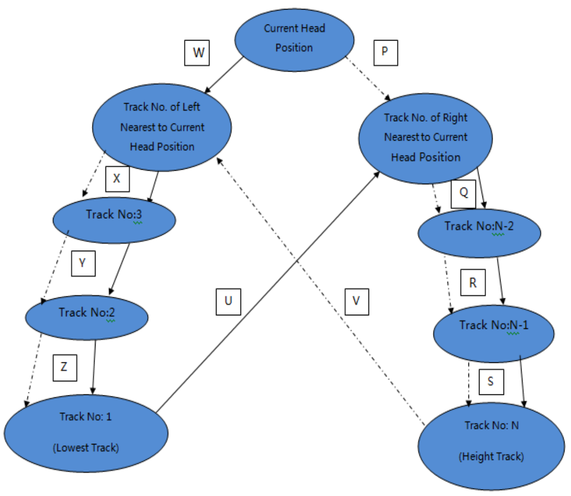


Fig3.1: Proposed method path flow

Path 1: (W+X+Y+Z+U+Q+R+S)
Path 2: (P+Q+R+S+V+X+Y+Z)
The X, Y, Z, Q, R and S are common in both paths
So, the movement depends on W and U or P and V.

If (W+U< P+V)
The path 1 says the lower movement.
Else
The Path 2 says the lower movement.

If path1 is selected then head moves and reached sequentially from its current position to the lowest block number in backward direction and again head moves and reached sequentially from its lowest block number to the highest block number in forward direction of those blocks which is not visited.
If path2 is selected then head moves and reached sequentially from its current position to the highest block number in forward direction and again head moves and reached sequentially from its highest block number to the lowest block number in backward direction of those blocks which is not visited.

3.1 STEPS OF PROPOSED SCHEDULING ALGORITHM

Step1: Start
Step2: Initialization an Array A[],Current Head Position H.
Step3: Read cylinder numbers and put it into an array and read current head position.
Step4: Sorting in ascending order of all cylinders input blocks by using any sorting method.
Step5: Find the value of path1 and path2.

i. path1= (current head position - cylinder no of left nearest to current head position) + (Cylinder no of right nearest to current head position - lowest cylinder).
ii. path2= (Cylinder no of right nearest to current head position - current head position) + (largest cylinder - cylinder no of left nearest to current head position).

Step6: If (path1 < path2)

Head moves and reached sequentially from its current position to the lowest block number in backward direction and again head moves and reached sequentially from its lowest block number to the highest block number in forward direction of those blocks which is not visited.

Else

Head moves and reached sequentially from its current position to the highest block number in forward direction and again head moves and reached sequentially from its highest block number to the lowest block number in backward direction of those blocks which is not visited.

Step7: End

3.2 GRAPHICAL REPRESENTATION OF PROPOSED ALGORITHM

Concider , for example a disk queue with requests for I/O to blocks on cylinders : Queue(0-199) :36,80,120,10,15, 40,188,150 ,168. This schedule is diagrammed in figure 3.

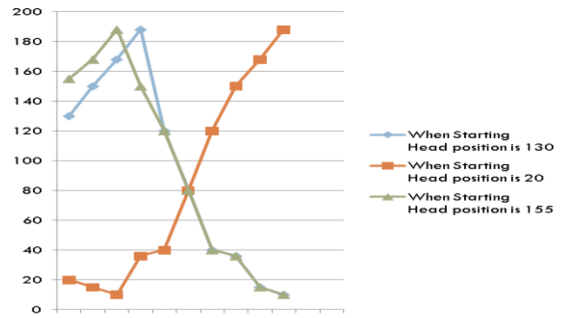


Fig3.2:Head movement of Proposed algorithm

3.3 MATHEMATICAL MODEL OF PROPOSED ALGORITHM

Let ,

Path1=P1
Path2 =P2
Head position=Hp
Request queue=a[]
No of request=n
Total Head Movement=Hm
Average Head movement=Ahm
Track position from Current Head Position to Right Nearest =rp
Track position from Current Head Position to Left Nearest =lp

$$p1 = Hp - a[lp] + a[rp] - a[1]$$

$$p2 = a[rp] - hp + a[n] - a[lp]$$

$$Hm = \begin{cases} p1 + a[lp] - a[1] + a[n] - a[rp];\ p1 < p2 \\ p2 + a[n] - a[rp] + a[lp] - a[1];\ p1 > p2 \end{cases}$$

$$Ahm = Hm / n$$

4. RESULTS ANALYSES

Table4.1: Comparisons table (for Total head movement)

S L. N O.	Name of Algorithm	Number of head movement(When starting head position is 20)	Number of head movement(When starting head position is 130)	Number of head movement(When starting head position is 155)
1	FCFS	49.34	58	60.78
2	C-SCAN	43.67	43.11	43.67
3	SSTF	**20.89**	28.44	24.55
4	C-LOOK	39	38.44	39
5	LOOK	**20.89**	33.11	35.89
6	SCAN	23.11	35.33	38.11
7	**New**	**20.89**	**26.22**	**23.44**

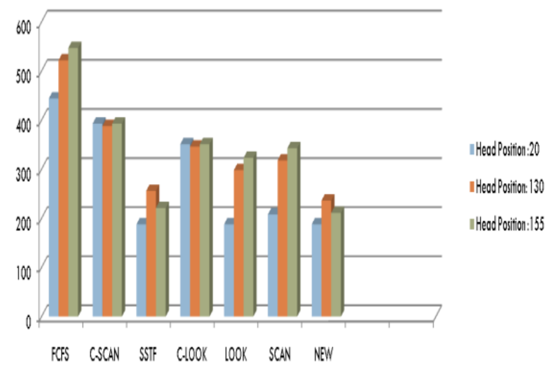
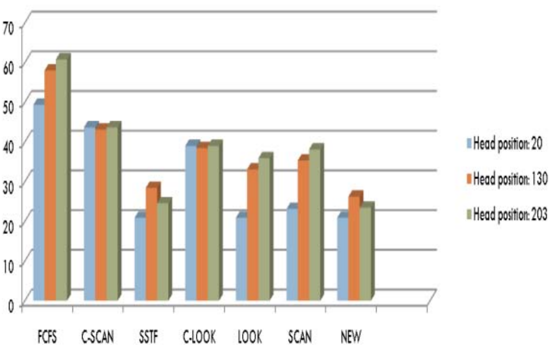


Fig4.1: Comparisons graph (for Total head movement)

Table4.2: Comparisons table (for Avg. head movement)

SL. NO	Name of Algorithm	Number of head movement(When starting head position is 20)	Movement(When starting head position is 130)	Movement(When starting head position is 155)
1.	FCFS	444	522	547
2.	C-SCAN	393	388	393
3.	SSTF	188	256	221
4.	C-LOOK	351	346	351
5.	LOOK	188	298	323
6.	SCAN	208	318	343
7.	New	188	236	211



5. LIMITATIONS:

1. Sometime number of head movement is equal to SSTF or LOOK scheduling.

2. When input blocks are stay in ascending order without sorting and head stay in starting block then FCFS is best. But in dynamic allocation of cylinder it is almost impossible.

3. It is not applicable for priority based request.

6. CONCLUSION

In conclusion, we propose an efficient, universal, low-maintenance disk I/O scheduling scheme that can automate the manual configuration and selection of disk schedulers. The scheduling scheme can learn about workloads, file systems, disk systems, CPU systems, and user preferences. From the above experiment and comparison of proposed algorithm with existing algorithm it is clear to us that the proposed algorithm reduces head movement and decide best route. It is new era for next generation.

REFERENCES

[1] Operating System Principles(6th edition) Abraham Silberschatz, Peter Bare Galvin, Greg Gagne.

[2] Mordern operating system(2nd edition) Andrew S. Tanenbaum .

[3] Operating Systems:A Concept-based Approach(2E) D.M. Dhamdhere.

[4] Sandipon Saha, Md. Nasim Akhter, Mohammod Abul Kashem.A New Heuristic Disk Scheduling Algorithm. International Journal of Scientific & Technology Research Vol. 2, Issue 1, January 2013.

[5] Reddy,A.L.N., Wyllie,J., and Wijyaratne, K.B.R. disk scheduling in a multimedia I/O system. ACM Trans. Multimedia Comput. Commun. Appl. 1, 1(2005), 37-59.

[6] SCSI architecture model-3(SAM3).Tech. Rep. T10 project 1561-D ,revision 14,International for information Committee Technology standards (INCITS),T10 Technical Committee. Sept. 2004.

[7] Chang, R., Shih, W., And Chnag , R. Real-time disk scheduling for multimedia application with deadline-modification-scan scheme. International Journal of Time-critical Computing System 19(2000),149-168.

[8] Dees,B. Native Command queuing-advanced performance in desktop storage.Potentials,IEEE 24, 4(2005),4-7.

[9] Frank,H .Analysis and optimization of disk storage devices for Time-sharing system. J ACM 16, 4(1969),602-620.

[10]Huang, Y., And Huang, J. Disk scheduling on multimedia storage servers. Computers, IEEE Transactions on 53, 1(2004), 77-82.

[11] Reddy,A.L.N.,And Wyllie,J. Disk Scheduling in a multimedia I/O system. ACM, pp.225-223.

[12] Ruemmler, C., And wiklkes,J. An introduction to disk drive modeling.Computer 27, 3(1994),17-28.

[13] C. Ruemmler and J.Wilkes. An introduction to disk drive modeling. *IEEE Computer*, 27(3):17–29, March 1994.

[14] Seaman, P.H., Lind, R. A., And Wilson,T.L.An analysis of auxiliary-storage activity. IBM System Journal 5, 3(1996), 158-170.

[15] Seltzer,M., Chen, P., And Ousterhout, J. Disk scheduling revisited, Jan . 1990.

[16] Yu, Y . J.,Shin, D. I., Eom,H., And Yeom,H.y. NCQ vs.I/O scheduler: Preventing unexpected misbehaviors. Trans. Storage 6, (2010), 1-37.

[17] Zhu, Y. Evalution of scheduling algorithms for real-time disk I/O. Tech.rep.,Department of cpmputer science and Engineering,University of Nebraska,

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Design an efficient MAC Protocol for Cognitive Radio Systems using Game Theory

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***Abstract*—In this paper we consider power distribution between cognitive users which is the most crucial problems in cognitive radio systems. To ensure perfect distribution game theory is applied. Conflicting users who are trying to access the same channel are detected first. Targeted signal to noise ratio (SNR) is achieved by iterative fashion which determines the signal strength. Power is calculated for each user by convergence theorem. Power is distributed in such way that it increases system utility. At last Nash equilibrium point is formulated for ensuring the avoidance of selfish behavior of cognitive users. Experimental results show that the proposed strategy achieves the ability to reduce the power consumption in order to increase the system utility.**

Keywords: Cognitive Radio, Conflicted players, Power Distribution, Nash equilibrium, Game Theory.

I. INTRODUCTION

Spectrum scarceness is one of the major challenges that the present world is facing. The efficient use of existing licensed spectrum is becoming most critical because of the growing demand of the radio spectrum. To solve this problem, cognitive radio (CR) has emerged as a leading technology because it can intelligently sense an unused spectrum without creating any harm to authorized users called primary users. In cognitive radio networks the secondary users can borrow the unused spectrum from the primary user for some time but this must be done by creating no harm to the primary user's communication.

In cognitive radio network there may be multiple secondary users. If there are fixed numbers of channels that are smaller than the total numbers of users then channel distribution among multiple users create severe problems. Suppose a channel is requested by many secondary users. Now the cognitive radio network will have to make a decision to accommodate the secondary user, who is requesting for utilizing the spectrum, specifying its application needs. This Spectrum allocation decision requires some factors or parameters like operating frequency, data rate, transmitting power, modulation scheme, signal bandwidth, error rate etc. for consideration of distribution. To mitigate this problem game theory is applied in this area that perfectly select the proper way of channel distribution.

In game theory, every user is considered as a player and they select their proper role in the channel distribution game. The players who are trying to access the same channel, are called conflicting players. If there is one player in each channel then there is no problem in distributing the channels. But when several players try to access a single channel it is essential to identify the right player who maximizes the spectrum utilization.

Figure 1 represents an example of conflicted players. Here P represents cognitive users (players) and C represents channels. Fig. 1 shows a motivating example where nine players (p1, p2,…,p9) are trying to access five channels (c1, c2,…, c5). Assume that the players first measure the channel-gain in all the available channels and then try to access the channel which has the maximum channel-gain. In fig 1 we see that in channel 2, there are three conflicting players (p6, p8, p9). Because, in this channel, the channel gain for all of these players are maximum among the channels. In channel 3 and 4, there are two conflicting players and in channel 1 and 5 there are only one player. The problem here is to allocate channels 2, 3 and 4 to the best one among the conflicting players so that the overall spectrum utilization is maximized.

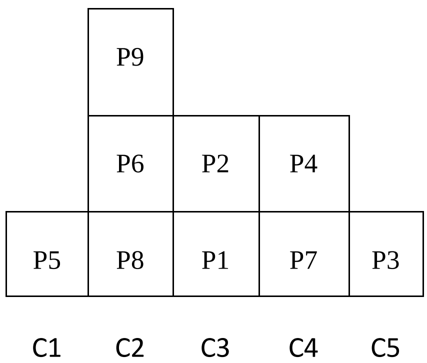

Figure 1. Channels of conflicted players

In this dynamic spectrum sharing case power distribution between conflicted players plays a vital role. A power distribution strategy is used and a utility function is introduced in [3]. A two- step power distribution algorithm based on game theory is proposed in [4]. A Medium Access Control (MAC)

Protocol based on price - based resource allocation algorithm is proposed in [5] for improving system throughput by reducing the average transmission power. A price-based iterative water-filling (PIWF) algorithm is proposed, which allows users to converge to the Nash Equilibrium (NE) that satisfies game theory. On the basis of stated research the power distribution problem between cognitive users is introduced in this paper. First the conflicted users are calculated manually in any channel. Then optimal SNR is calculated in iterative fashion. At last Nash equilibrium point is formulated with the presence of SNR. At this point the optimal power would be found for all conflicted players.

The paper is organized as follows. In section 2, Methodology is represented. Game theoretic model is presented in section 3. In section 4, simulations results are shown. Conclusion is represented in section 5.

II. METHODOLOGY

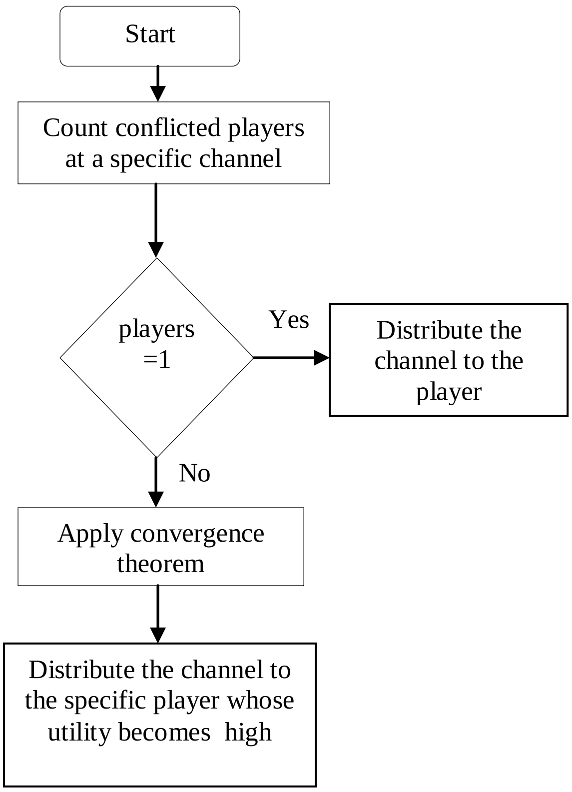

Figure 2. Flow chart for the overall methodology

Figure 2 represents the methodology of the system. At first the system's vital role is to find out the conflicted players at a specific channel. When it exceeds one, it will calculate the convergence point for each player that will represent the power consumption for each player. After this, the system will find out the utility and based on the utility, channel will be distributed.

III. SYSTEM MODEL

We consider a cognitive radio network with N radio transmitter-receiver pairs distributed randomly in an area. We assume that in this area, there are K frequency channels that are available for transmission between the radio pairs. In the game theoretic model each transmitter is considered as a player. Here all players, individually, try to access the specific channel in which their respective channel-gains are maximum.

Now our role is to find out the equations of those players whose channel-gains are maximum in the same channel. Assume that Q transmitters use the same channel c_0. They have power,

$$P : \left(p_0, p_1, p_2, \ldots\ldots\ldots\ldots, p_Q\right)$$

Here, p_i represents the power at the ith transmitter. Where

$$i = 1,2,3,\ldots\ldots\ldots, Q$$

The expression for SNR at receiver i is,

$$SNR = \frac{g_{ii}\,p_i}{\sum_{j=1,\,j\mp i}^{Q} g_{ij}\,p_j + n_i}\eta \qquad (1)$$

$$g_{ij} = \text{link gain}$$

$$n_i = \text{noise power at receiver i}$$

Now our task is to select power level that exactly meet target SIR. It is denoted by γ_0. SNR maintains some criteria. That is,

If $SNR > \gamma_0$, use too much power

If $SNR < \gamma_0$, packets cannot be received correctly

So we can reach in the conclusion that transmitter i is supported if :

$$SNR_i \geq \gamma_0 \text{, here } \gamma_0 = \text{target } SNR$$

When the targeted SIR is obtained then the equation of power can be stated as :

$$p_i \geq \gamma_0 \left(\sum_{j=1,\,j\mp i}^{Q} \frac{g_{ij}\,p_j}{g_{ii}} + \frac{n_i}{g_{ii}}\right) \qquad (2)$$

Denote $\frac{g_{ij}}{g_{ii}} = h_{ij}$; $\frac{n_i}{g_{ii}} = \eta_i$

The users adjust their power to meet or exceed target SNR. So at last the minimized equation is obtained as-

$$p_1 \geq \gamma_0\left(h_{12}\,p_2 + \eta_1\right) \qquad (3)$$

$$p_2 \geq \gamma_0 (h_{21} p_1 + \eta_2) \quad (4)$$

So the Minimum power solution is:

$$p_1 = \gamma_0 (h_{12} p_2 + \eta_1)) \quad (5)$$

$$p_2 = \gamma_0 (h_{21} p_1 + \eta_2) \quad (6)$$

Here equations 5 and 6 are our concluded state in which players 1 and 2 are conflicted. It implies that here the channel gain is maximum for both players. So game theory is applied here to formulate the minimized solution. At last Nash equilibrium point is calculated by convergence theorem. The same process is applied for multiple players who are trying to access the same channel in order to be maximum channel gain in that channel.

The utility of user j,

$$u_j = \frac{ER}{p_j}(1 - e^{-0.5\gamma_j})^L \quad (7)$$

E= energy content of battery

L= packet length

R= data rate

p_j = power at the j^{th} transmitter

A reverse relation exists between power and utility. From equation 7, it is noticeable that if power increases then the user utility decreases. As our vital role is to increase the payoff of the system, so we have to increase utility. If it is somehow possible to decrease the power consumption, then the utility will be easily increased.

IV. SIMULATION RESULTS

In this section we provide our experimental results and the comparison between the existing work. Here we prove the effectiveness of our proposed model. In our experiment we consider the available channels are less than the cognitive users, so that the channels become conflicted between users. The experimental parameters are listed in table 1.

To find out the target SNR is the initial task to our experiment. The number of iterations increases when targeted SNR becomes larger during power calculation.

Figure 3 shows the Experimental results of SNR versus iterations of proposed and existing work [4]. It represents the number of iteration required to reach target SNR for two different models. From this result we can conclude in the discussion that in order to reach in the targeted SNR, initially our proposed model requires a little bit more iterations. But at the end the required iterations for desired SNR is quite similar.

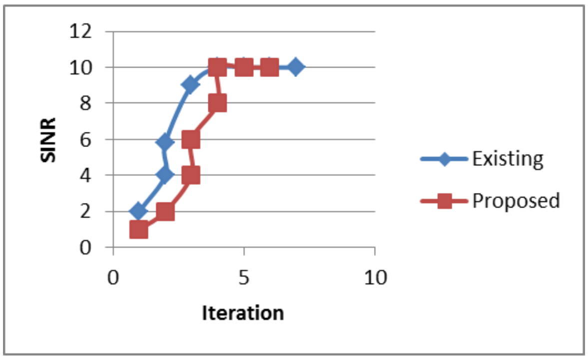

Figure 3. SNR versus iteration

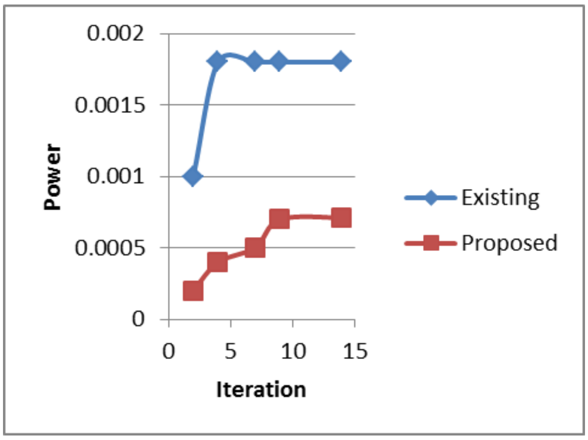

Figure 4. power versus iteration when SNR=10

Power versus iteration curve is shown in Figure 4. It also depends on signal to noise (SNR). Power consumption increases with the increasing order of SNR. Our goal is to keep the SNR in a tolerable limit so that the iterations and the power consumption become low.

In Figure 4 we notice that in our proposed model converging steps (number of iterations) are larger than the existing model [4] to reach in a steady state. But here power consumption is lower than the existing model [4]. As our target is to reduce power consumption, so in this case our proposed model is better than the existing model.

Utility varies reversely with the increasing numbers of conflicted players as stated in equation 6. This relationship implies that the consequences of increasing power consumption decreases utility. Therefore, we can come into conclusion that with the increase of the number of conflicted users, the number of iteration increases and the amount of power is also increased up to a certain stage. Consequently, the utility decreases when the number of conflicted players increases. Figure 5 shows the number of conflicting users versus utility. As we expected, the utility decreases with the increase of the number of conflicting users.

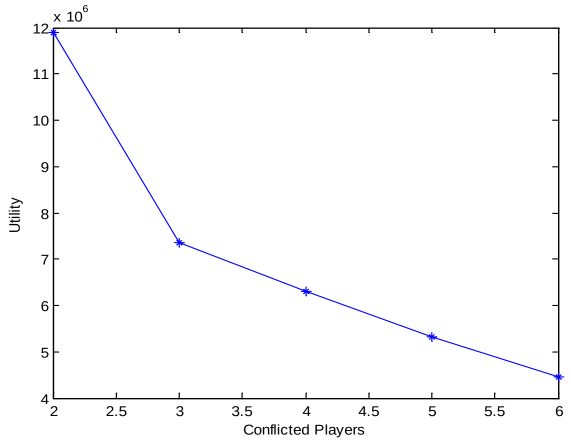

Figure 5. Utility versus conflicting players

Table 1. Experimental Parameters

TRANSMISSION RATE R :	10^3 B/S
LENGTH OF INFORMATION BITS L :	30
NOISE POWER N_0 :	10^{-4} W
MAXIMUM TRANSMISSION POWER :	1W

V. CONCLUSION

The Channel distribution based on power consumption is one of the major challenges in cognitive radio networks. Power sources become more lasting only when the power consumption is low. Here we have reduced power consumption efficiently. Converging steps increase with the increasing order of signal to noise ratio. This ultimately increases power consumption. We have reduced power consumption by slightly increasing in the number of converging steps.

If our proposed model is applied when multiple players try to access the same channel, it will pick up the exact player to give control of the channel with the cooperative manner. So at the end point it can be concluded that the proposed model is a somehow favorable solution for this power distribution problem in a cognitive radio (CR) network. It gives the better solution among the available solutions in existing literature.

When the number of conflicted players becomes very large then sometimes the convergence may not come. But it may happen only in the rare case. Somehow it can be improved in future.

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REFERENCES

[1] I. F. Akyildiz, W. Y. Lee, M. C. Vuran, and S. Mohanty, “A survey on spectrum management in cognitive radio networks," IEEE Communications Magazine, vol. 46, no. 4, pp. 40-48, 2008.

[2] J. Márk Félegyházi, Member, IEEE, Mario ˇ Cagalj, Member,IEEE, ” Efficient MAC in Cognitive Radio Systems: A Game-Theoretic Approach," IEEE transactions on wireless communications, Vol. 8, No. 4, April 2009.

[3] I. F. Wang, M. Krunz, and S. Cui. Price-based spectrum management in cognitive radio networks. IEEE Journal of Selected Topics in Signal Processing, 2(1): 74-86, 2008.

[4] Rui YANG†, Yibing LI, Fang YE, “A Two-Step Power Distribution Algorithm Based on Game Theory in Cognitive Radio Networks," *Journal of Computational Information Systems* 7: 10 (2011) 3585-3590.

[5] Fan Wang,Marwan Krunz,and Shuguang Cui, "Spectrum Sharing in Cognitive Radio Networks Technical Report TR-UA-ECE-2007-1.

[6] Y. C. Yang, J. Li, W. Li, and D. Chen. “Power allocation based on noncooperative game theory in cognitive radio,” Journal of Xidian University,36(1): 1-4, 2009.

[7] S Cheng, Z. Yang, and H. Zhang.Novel,” power control game algorithm for cognitive radios,” Journal of Communications, 28(3):100-106,2007.

[8] C. U. Saraydar, N. B. Mandayam, D. J. Goodman, “Efficient power control via pricing in wireless data networks,” IEEE Transactions on Communications, 50(2): 291-303, 2000

Dimensionality Reduction and Cluster center Selection: An Efficient Scheme for High dimensional dataset Clustering.

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***Abstract*— In the advance technology data volume with many objects and dimensions is increasing day by day. At high dimensional dataset traditional clustering algorithms do not perform well. With increasing dimensionality some traditional algorithms produce local best possible results. There are two major issues of many partitioning clustering algorithms; one is relevant feature selection and other is selection of optimal initial clusters center. K-means is a popular clustering algorithm. But it is unable to cluster a high dimensional dataset for "curse of Dimensionality" problem .It has a major problem "to select initial centriod ". In this paper, we have proposed a technique for selecting winner node of KSOM algorithm which reduces most relevant dimensions of data set. And we modified k-means algorithm for selecting initial centriod to get better cluster. When, we compared the results of proposed technique with existing techniques, our modified KSOM and K-means produce better cluster. We experimented with an IRIS data set and Milk dataset, its performance was compared with other clustering algorithm for siddheswar index, quantization errors and topographic errors.**

Keywords—Data mining, k-means clustering, Kohonen Self Organizing Map(KSOM), curse of Dimensionality, Clustering.

I. INTRODUCTION

Clustering is the data mining process, which creates the clusters of objects where objects having similar property, which is an unsupervised learning algorithm. To make cluster some popular distance function, e.g., Euclidean distance, Manhattan distance equation is used to compute the similarity between two objects. Minimum Euclidean distance represents the objects are more similar with other objects. Traditional clustering methods K-Means does not work well for calculating similar objects.

There are two types of clustering algorithms. They are partitioning algorithm and Hierarchical algorithm. Partitioning algorithms start with the random selection of *k* objects, from the data set. These initial clusters are refined to minimize the sum of squared error among the clusters. [1]

To create a tree structure of the objects in the data set Hierarchical clustering algorithm is used that is called dendrogram. These methods create k clusters by cutting the dendrogram at the specific level. The traditional clustering algorithms create clusters based on the similar space, for this reason such type of algorithm fail to work on high dimensional data set. On the other hand the partitioning clustering algorithms choose the initial clusters centers randomly, so the efficiency is not good and the exact cluster is not found, they produce different cluster results for optimal solution. [2]

In the data mining techniques there has a major problem i.e Dimensionality reduction. There are two distinct methods can be reduce the dimensionality. That is feature transformation (FT) and feature selection (FS).DIMENSION reduction means that to reduce the size of the representation of the data objects. That's why computational cost is less in the next steps.

Principal Components Analysis (PCA) or Singular Value Decomposition (SVD) is used for Feature transformation. They make some linear transformation on high dimensional data set and produce low dimensional data set for further processing [3]. On the other hand, Feature selection is a processing step that is used to remove unwanted or noisy dimension of the data set .It also selects the most relevant dimensions of the data set based on ranking of the dimension [4]. Dimension reduction is essential for production of better cluster.

There are two well-known methods for unsupervised feature selection; filter methods and wrapper methods. These feature selection techniques provide better clusters. But computational cost is high.

In this paper, we proposed a technique which was used for dimension reduction of data set named modified KSOM algorithm and for better clustering we used modified K-means algorithm to select efficient initial clusters centers. Our proposed technique consists of three steps: dimension reduction step done by modified KSOM, initial centriod selection step done by modified k-means algorithm and the final step i.e final cluster is done by used in the k-means algorithm to obtain final clusters. Our approach is highly adaptive to any kind of clusters.

Rest of the paper is organized as follows. Section II summarizes the previous relevant work. In section III presents the experimental steps. Original KSOM and modified KSOM is described in the section IV. In the section V describes original k-means and proposed initial centeriod k-means algorithm. Section VI describes the experimental result. We present some comparison criteria and compare the results with other existing Clustering algorithms in section VII. Finally, Section VIII gives the conclusion and advance scope of the Work

II. LITERATURE REVIEW

Initial centroids selection is strongly hard in the original k-means algorithm for randomly choose. Due to unrelated selection of centroids leads to many unlike resulting clusters. As a result the accuracy and quality of k-means clustering algorithm is not good. To progress that criteria of k-means several different methods have been proposed over the last few years [5].

Principal Component Analysis and Multidimensional Scaling are used for Dimensionality reduction. They describe handling of linear data. On the other hand Linear Embedding (LLE) and Isometric Feature Mapping (Isomap) [6] presents dimension reduction of nonlinear data with some type of topological manifold. However, LLE and Isomap both are unsuccessful in some other type of data, e.g., sphere or torus data set.

There have been many works done on civilizing the performance and efficiency to clustering of high dimensional datset. Dash, et al and Hs Behera et al [7] was proposed a hybridized K-Means clustering approach for high dimensional data set where PCA ,Canonical Variate analysis and Genetic Algorithm was used for dimensional reduction. To find out the initial centroids a new method is employed that is calculating the mean of all the data sets and then divided this value into k different sets in ascending order.

Arai et al. was proposed a hierarchical algorithm to determine the initial centers for K – means [8]. An algorithm called CCIA was presented by Khan et al. [9], in that paper to determine the similarity of data objects first compute the mean and standard deviation of each dimension of data set and then partition the data by normal curve into some clusters .But the accuracy was not good.

Juha Vesanto ,Esa Alhoniemi and.J. Vesanto [10] was proposed Self Organizing Map for clustering..In that work they proposed the SOM based data visualization to visualize the data's using SOM. But time complexity is so high for clustering high dimensional dataset. However, Apolloni et al. [11], present a method called Boolean Independence Component Analysis (BICA), which fetch Boolean bits with minimal joint entropy and consistence assignment from full dimensions, then it produces a vector of Boolean variable with unique assignment. This obligation is directly proportional to relevance of the feature and same assignment does not assign to another classification level.

Kathiresan V. et al. in was proposed on Z-Score Ranking method to select initial centroids for high dimensional dataset .The author proposed to calculate the Z-Score of each data point in the data set and then sorting the data points based on the Z-Score values. After that the mean value of each subset is calculated and finally any close to value act as initial centroid.By this way experimental result shows both accuracy and time complexity. [12]

For feature evaluation and selection Hu et al. [13], proposed a technique to the concept of classification loss and neighbourhood margin. Based on soft margin support vector machines and neighbourhood rough set it determines the classification accuracy of feature subsets.

Non-linear dimension reduction technique called multi-layer isometric feature mapping (ML - Isomap) was proposed by Liu et al. [14].The author presented that the cluster is automatically produce by using rushes editing. To improve estimation accuracy in High-dimensional data sets Sugiyama et al. [15] proposed Direct Density-ratio estimation with Dimensionality reduction (D3) method.

Constructive Approach for Feature Selection (CAFS) was proposed by Kabir et al. It is based on the concept of the wrapper approach and sequential search strategy. By using correlation information the method determine the feature selection. It also uses three layered feed-forward neural network with the neural network architecture. But the time complexity of this algorithm is also high.[16]

There have been many works done on improving the accuracy of high dimensional dataset clustering. An algorithm by Barakbah et al. [17] was presented a method for selection of initial clusters centers. The pillar designates approach is used in this algorithm. But the algorithm has a problem to reduce unwanted dimension.

III. PRELIMINARIES :

A. Original KSOM

The Self-Organizing Map is a very popular artificial neural network (ANN) algorithm based on unsupervised learning. The SOM is used in various data mining tasks. Vector quantization and projection (like Sammon's mapping)is the very beneficial properties. The Kohonen SOM was first introduced in 1982 by Tuevo Kohonen, a professor emeritus of the University of Helsinki. It is basically a single-layer feed forward network. (Input layer to feature map).

Teuvo Kohonen writes "The SOM is a new, effective software tool for the visualization of high-dimensional data. It converts complex, nonlinear statistical relationsihps between high-dimensional data items into simple geometric relationships on a low-dimensional display. As it thereby compresses information while preserving the most important topological and metric relationships of the primary data items on the display, it may also be thought to produce some kind of abstractions." [15].

The nework has two layers i.e input layer acts as a distribution layer and output layer. Each unit in the 1st layer takes on the value of the corresponding entry in the input pattern and the 2nd layer units then sum their inputs and compete to find a single winning unit. By observing the input patterns, KSOM reorganizes those by clustering similar patterns into groups. The weight nodes are initialized with random numbers. The ``winning" mapping node is defined as that with the smallest Euclidean distance between the mapping node vector and the input vector The weight vector of the unit is closest to the current object becomes the winning or active unit. During the training stage, the values for the input variables are gradually adjusted in an attempt to preserve neighborhood relationships that exist within the input data set. After all of the input is processed (usually after hundreds or thousands of repeated presentations), the result should be a spatial organization of the input data organized into clusters of similar (neighboring) regions.[15]

B. *Orginal K-Means Algorithm*

K-means algorithm is very popular algorithm for clustering. The original k-means clustering algorithm describes in below. There are two phases: the first phase is to define k numbers i.e cluster number. The subsequently contains initial centriod point it is choose by randomly. In this algorithm Euclidean distance is generally considered to determine the distance between data points and the centroids. When all the points are included in some clusters, the first step is completed and an early grouping is done. At this point we need to recalculate the new centroids, as the inclusion of new points may lead to a change in the cluster centroids.

Once we find k new centroids, a new binding is to be created between the same data points and the nearest new centroid, generating a loop. As a result of this loop, the k centroids may change their position in a step by step manner. Eventually, a situation will be reached where the centroids do not move anymore. This signifies the convergence criterion for clustering.

But the main drawback of this algorithm is initial centriod selection. For selection of initial centriod is randomly. The time complexity is high also accuracy is not good. The time complexity of this algorithm is O(nkl) where n is the number of data points, k is the number of clusters and l is the number of iterations [7]

IV. PROPOSED METHOD

A. *Modified KSOM*

Input:

//Set up input layer, total number of input neurons, I = Input P*Input Q.

// Set up output layer. Total number of output neurons, J =Output P * Output Q

//Initialize connection weights (randomize) between input layer neurons and output layer neurons, Wij

//Set initial learning rate parameter, α (between 0.2 to 0.5)

//Set total number of iterations, T

//Start with iteration t = 0.

Output: A set of K resultant clusters.

The Pseudo code of the modified KSOM algorithm

Step 1. The network is fully connected that is all nodes in input layer are connected to all nodes in output layer.

Step 2. Apply the first input of the KSOM and Compute the winning neuron (jc) in the output layer which is the minimum Euclidean distance .The Euclidean distance as follows

Modified portion of the KSOM algorithm

$$E_j = \sqrt{\sum_{i=0}^{i=n-1} (w_{ij} - x_i)^2} \qquad .(1)$$

For winner neuron: Calculate the mean and standard deviation of the each part of all dimensions

$$\mu_X = \frac{\sum_{i=1}^{n} X_i}{n}, \sigma_X = \sqrt{\frac{\sum_{i=1}^{n} (X_i - \mu_x)^2}{(n-1)}} \qquad (2)$$

The winner neuron, jc:

$$j_c = \frac{(\mu_x + \sqrt{\sum_{i=0}^{n-1} (\mu_x - E_i)})}{\sum^{n} \sigma_i^2}$$

Step 3. Update weight for each connection i.e. for all neurons j and for all i:

$$\begin{aligned} w_{ij}(new) &= w_{ij}(old) + \Delta w_{ij}(new) \\ Where, \Delta w_{ij}(new) &= \alpha_t (x_i - w_{ij}(old)) \end{aligned} \qquad (4)$$

Step 4. Update learning rate α such that:

$$\alpha_t = \alpha_0 (1 - \frac{t}{T}) \quad (5)$$

Step 5. Increase iteration t=t+1 until t=T

Step 6. Repeat with next pattern chosen randomly(Do Steps 2-6)

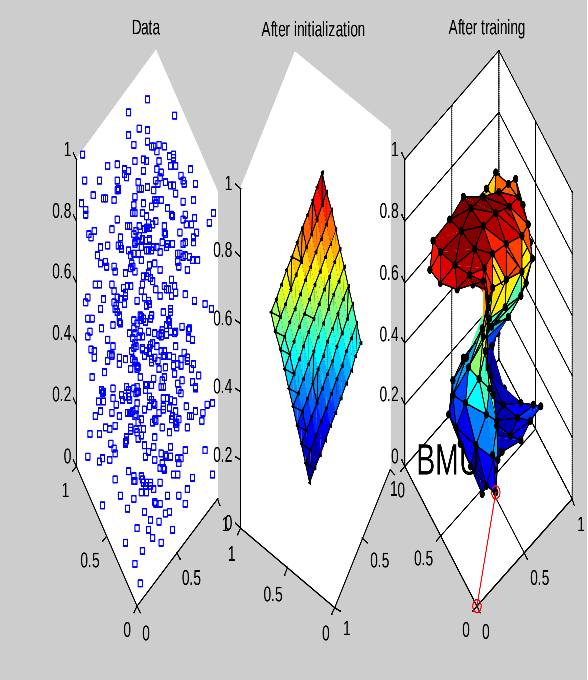

Figure 1. Efficent best matching unit(BMU) by proposed Algorithm.

B. Proposed Initial Centriod K-Means Algorithm

In the following method is used to find out the initial centriod of the dataset. For example a data set D is consists of n data such as d1, d2, d3,....., dn. Each data point of this set may contain multiple attributes such as di may contain attributes a1, a2, a3,..., am, where m is the number of attributes. In case of multidimensional attributes we set a load (0-1). After multiplying the load factor with each attribute we calculate the Euclidean distance among all the data objects and then sort this value by merge sort. ***Divide*** the datasets into k subsets and calculate median value .Identify the initial centroids whose Sum value (SV) is nearby median value of K-means algorithm.

.

The Pseudo code for the algorithm is as follows.

Input:

D = d1, d2.......dn // set of n data items

L // set of load for data points.

Output: A set of Initial centriod.

Steps:

Step 1. $data[i] = d_1, d_2, d_3, ..., d_n$ for $i = 1$ to $n-1$

While $(j <= n)$

Step 2. Multiply the load with data value.

$$SumValue[j] = load[i] * data[i] \quad (6)$$

a) ***For*** $j = 1$ to $n-1$ and ***For*** $i = 1$ to $n-1$

b) $load[i] = (0-1)$

Endfor //Where d = the data value of the attributes, data =array of data value, n = number of attributes and l = load to multiply to ensure fair distribution of cluster .

Calculate the Euclidean distance among all the data objects.

$$\|E(j)\| = \sqrt{\sum_{i=0}^{i=I-1} (Sumvalue[j] - Sumvalue[j+1])^2} \quad (7)$$

Step 3. Take the sorts possible data point of the data object for each data subsets by merge sort ;

$$\|E(j)\| = SORT\|E(j)\| \quad (8)$$

Step 4. Divide the datasets into k subsets;

Step 5. **For** each subset

a) Calculate median value

b) Select a initial centroid whose SumValue (SV) is nearby median value.

Endfor

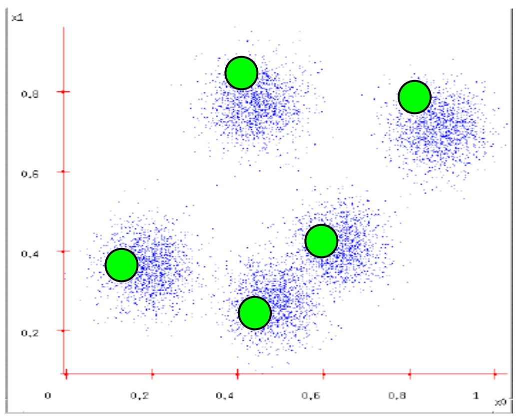

Figure 2. Optimal selection of initial centroids in proposed method .

V. EXPERIMENTAL RESULTS

For clustering of high dimensional dataset we take two standard dataset for our experiment: milk data set and Iris data set . Some existing clustering algorithm such as principal component analysis based k-means, Sammon mapping canonical vector analysis and our modified KSOM and modified K-means .We judge against the results using some well known comparison criteria.

A. *Experiment 1: Milk Dataset*

Milk data set contain 25 mammals and 5 ingredients Water, Protein, Fat, Lactose, and Ash of their milk, where mammals are represented *(M_1,M_2.........M_{25})* and ingredients of their milk are considered as the objects *(o_1,o_2......o_5)* of the data set. Here dimension reduction and number of clusters is get from KSOM. So after dimension reduction and get the cluster number we have to cluster these mammals into k=2 clusters based on the similarity in their ingredients of milk. Visualization of clusters using SOM is shown in figure 3.

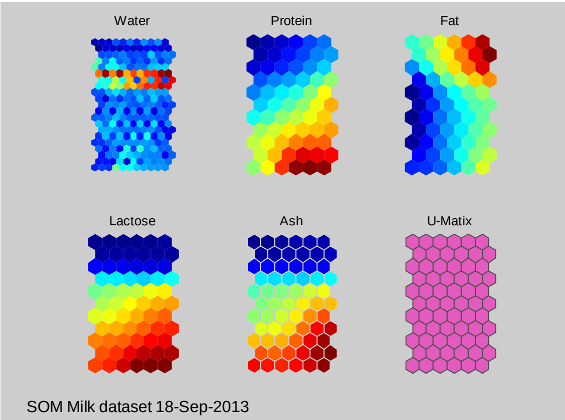

Figure 3. Visualization of clusters for Milk dataset

When we apply milk dataset for dimension reduction in the KSOM algorithm then dimension will be reduce by Modified KSOM porsion. After that weight will be updated and get preliminary cluster no (K=2) classes. Then for better clustering we applied modified K-means algorithm. Here, the initial centriod of the cluster of 2 groups are (1.584, 2.354, 2.1678) and (1.53, 9.15, 4.41, 5.05, 9.34, 2.00).These objects are referred as initial centers of clusters for C_1 *and C2* respectively. At that time in the final step, initial cluster centers obtained in used in the k-means to find optimal clustering of data set. The mammals contained in these clusters are Cluster1= *(M_{24}, M_{25}, M_8, M_9, M_{10}, M_{11}, M_{14}, M_{16}, M_{17}, M_{18}, M_{19})* and Cluster2= *(M_{20}, M_{21}, M_{22}, M_{23}, M_1, M_2, M_3, M_4, M_5, M_6, M_7, M_{12}, M_{13}, M_{15})* respectively, Clusters are shown in figure 4

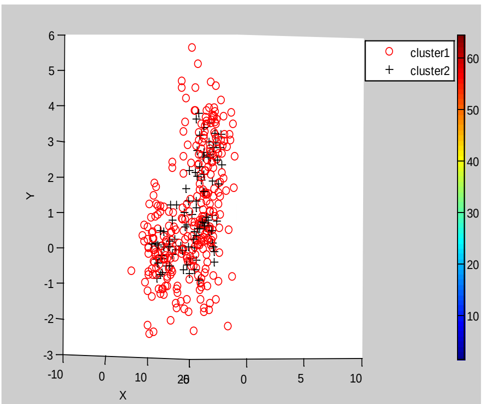

Figure 4. Clustering result of Milk Data Set by modified KSOM and K-means .

B. *Experiment 2: IRIS Dataset*

The experimental analysis is performed on an IRIS data set which we can get from UCI Repository of Machine Learning Databases. Actually, the data set consists of 50 samples of three species of Iris-flowers (a total of 150 samples) such that the measurements are width and height of sepal and petal leaves. The label associated with each sample is the species information: 'Setosa', 'Versicolor' or 'Virginica'. The data set contains 5 dimensions of three types of flower species. That is *((d1,d2......d5)*. Based on the length and width of sepal and petal we are to cluster these different flower species.

The KSOM algorithm will reduce the dimension of Iris dataset by Modified KSOM portion. Then we get preliminary cluster no (K=4) classes. To get exact cluster we applied modified K-means algorithm in the same dataset. Here, Clusters are shown in figure 5

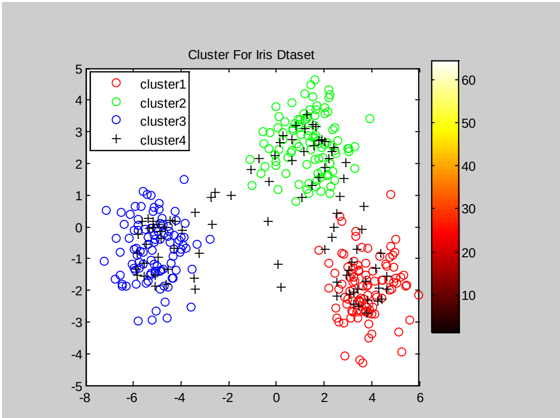

Figure 5. Clustering result of Iris Data Set by modified KSOM and K-means

At this time, we show the comparison of results of existing clustering techniques and modified KSOM, K-means algorithm on both milk data as well as Iris Dataset using three well known comparison points. The results of milk data set and Iris data set are shown in table 1, Figure 6 and table 2, Figure 7 respectively. Here, it is clearly found out that it is to have better performance in terms of all criteria. The results of existing techniques as well as our proposed technique then compare the results as shown in table 1 and table 2.

Table 1. PERFORMANCE COMPARISON OF MILK DATASETS

Algorithm	*Siddheswar Index*	*Quantization Error*	*Topographic Error*
PCA+K-means	0.0757	0.7756	0.2295
Sammon Mapping	0.0742	0.7345	0.2689
CVA	0.0731	0.6785	0.2526
Proposed method	0.0631	0.3813	0.0125

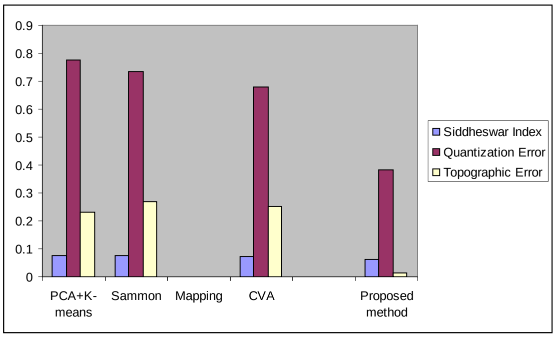

Figure 6. Comparison of the different criteria on Milk dataset

Table 2. PERFORMANCE COMPARISON OF IRIS DATASETS

Algorithm	Siddheswar Index	Quantization Error	Topographic Error
PCA+K-means	0.0364	0.8643	0.3287
Sammon Mapping	0.0326	0.8134	0.4321
CVA	0.0382	0.7678	0.2478
Proposed method	0.0234	0.3613	0.0125

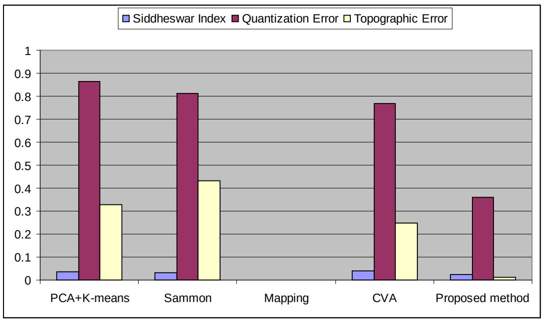

Figure 7. Comparison of the different criteria on Iris dataset

VI. CONCLUSION AND RECOMMENTATION

High dimensional data contains sea of noise and many irrelevant dimensions. Now a day, there are many applications of high dimensional dataset. So the results of the cluster should be exact. On the other hand, K-means algorithm is simple and popular. But it is unable to handle dimensional dataset and random selection of initial cluster centre. So it gives worse result. In this paper, we present a clustering technique, which provide the solution for both of the problems. Therefore, in order to get the local optimal clustering we applied modified KSOM for dimension reduction and finally we used an efficient approach for selection of initial centeriod of K-means algorithm .The comparative result is superior to the existing algorithms in all cases i.e Siddeswar Index, Quantization Error and Topographic Error. More effective feature detection technique will make the system more powerful. Advance work will be done to reduce the iteration of KSOM algorithm and to optimize $\eta(t)$.

REFERENCES

[1] Pearson, K. “On lines and planes of closest fit to systems of objects in space”, Philosophical Magazine, Series B,2(11), 559–572, 1901.

[2] Yedla, M. et al, ”Enhancing K means algorithm with improved initial center”, (IJCSIT) International Journal of

[3] Computer Science and Information Technologies, Vol. 1 (2) , pp- 121-125,2010

[4] Cox, T. F., and Cox, M. A. A., “Multidimensional scaling (2nd ed.). /CRC”, New York: Chapman & Hall, 2000.R.

[5] Richard O. Duda and Peter E. Hart, “Pattern classification and scene analysis”, Wiley, 1973

[6] Fahim A. M., et al, “An efficient k-means with good initial starting points”, Georgian Electronic Scientific Journal: Computer Science and Telecommunications, Vol. 2, No. 19, pp. 47-57,2009.

[7] Tenenbaum J., de Silva V. and Langford J., “A global geometric framework for nonlinear dimensionality reduction”, Science, 290(5500), 2319–2323, 2000.

[8] Behera, H. S. et al, “An improved hybridized k-means clustering algorithm(IHKMCA) for high dimensional dataset and its performance analysis” International journal of Computer science & Engineering,Vol-3 no-2,pp 1183-1190,2011

[9] Kohei Arai and Ali Ridho Barakbah, “Hierarchical Kmeans: an algorithm for centers initialization for Kmeans”**,** Reports of the Faculty of Science and Engineering, Saga University, Vol. 36, No.1, 2007.

[10] Khan, S.S. and Ahmad, A., “Cluster center initialization algorithm for K-means clustering”, Pattern RecognitionLetter, 2004.

[11] Siddheswar Ray and Rose H. Turi, “Determination of Number of Clusters in K-Means Clustering and Application in Colour Image Segmentation”, ICAPRDT,pp. 27-29, 1999

[12] Bruno Apolloni, Simone Bassis and Andrea Brega, “Feature selection via Boolean independent component analysis”, Information Sciences 179, 3815–3831, 2009

[13] Kathiresan V. and Dr. P Sumanthi, “An Efficient Clustering Algorithm based on Z-Score Ranking method,” *International Conference on ComputerCommunication and Informatics (ICCCI-2012)*, Jan. 10-12, 2012

[14] Qinghua Hu, Xunjian Che, Lei Zhang and Daren Yu, “Feature evaluation and selection based on neighbourhoods oft margin”, Neurocomputing 73, 2114–2124, 2010.

[15] Yang Liu, Yan Liu and Keith C.C. Chan, “Dimensionalityreduction for heterogeneous dataset in rushes editing”,Pattern Recognition 42, 229 – 242, 2009

[16] http://en.wikipedia.org/wiki/Self-organizing_map

[17] Md. Monirul Kabir, Md. Monirul Islam and Kazuyuki Murase, “A new wrapper feature selection approach using neural network”, Neurocomputing 73, 3273–3283, 2010.

[18] Ali Ridho Barakbah and Yasushi Kiyoki, “A pillar algorithm for kmeans optimizatioa n by distance maximization for initial centroid designation”, IEEE, 2009.

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Development of a Digital Weighing Machine Using 89S52 Microcontroller Architecture

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ABSTRACT
In a Digital Weighing Machine (DWM), the system goes through the steps (i) acquire weight using a precise load cell, (ii) normalizing the acquired raw weight by the gain and offset of the input subsystem, (ii) acquire rate, (iii) compute cost, and (iv) show the weight, rate and cost on display unit. This paper presents the implementation methodology of these steps using real hardware. A prototype meter was emulated using 89S52/HX710 architectures and MicroTalk-8051 Learning/Dev. System (Fig. 12), which was found to work within specifications.

KEYWORDS: Digital Weighing Machine, 24-Bit ADC, Calibration, Load Cell, FPGA.

1. INTRODUCTION

With the advent of microcontroller, high precision analog-to-digital converter and load measuring sensor, we are now in a position to transform our measurement technology from old-age analog domain to digital domain. Introduction of Digital Weighing Machine (DWM) has drastically improved the life style of general public by saving time, which would otherwise be spent in the manual calculation of cost.

1.1 Motivation for the Project

Now-a-days, in Bangladesh, almost every shop uses Digital Weighing Machine for the ***(i)*** accurate measurement of goods, ***(ii)*** rate entry from a keypad, and ***(iii)*** showing cost, weight and rate on bright display unit. These DWMs are imported items and cost huge foreign currencies. The design of the DWMs is based on proprietary FPGA (Field Programmable Gate Array) modules and as such imposes great difficulties to repair the faulty machines. The author has designed, developed and tested a Smart DWM based on MCU (not FPGA), which satisfies all the basic features of a DWM.

1.2 Design

The design of the DWM involves the transformation of a highly complex computing algorithm (Fig. 9-11) into 'Program Codes' and 'Hardware'. The program codes have taken care of ***(i)*** acquiring weight from load cell, ***(ii)*** converting the weight information into 24-bit data, ***(iii)*** calibrating the weight information to fit with the ideal response curve (Fig. 2) $y = mx$, ***(iv)*** acquiring rate from the keypad, ***(v)*** computing cost, and ***(vi)*** showing the Cost, Weight and Rate on bright display unit. The formula exclusively belongs to Bangalee intelligence and the product could not be easily copied. It is possible to add sophisticated feature like credit card payment with this machine in order to make it a standard export item.

1.3 Socio-Economic Impact

We have hundreds of shops in the country to use these Digital Weighing Machines. Therefore, there is a huge demand. Local production of these machines will save a lot of foreign currencies. There would be openings for jobs for engineers, technicians, workers and marketing people. There will also be opening for talented graduates to do research for the next models of low-cost DWMs using FPGAs.

1.4 Brief Functional Description of the DWM

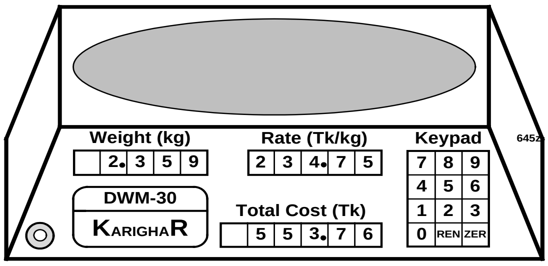


Fig. 1 Hypothetical View of Digital Weighing Machine

In Fig. 1, we have presented a hypothetical pictorial view of our proposed Digital Weighing Machine (DWM). Here is a brief description of the operational features of the DWM.

- At power up and without any load on the pane, all the display fields will show 00s.
- In case, the weight field shows any stray weight, the user may press the ZERO button to nullify the stray (tare/idle) weight.
- By default, the DWM keeps acquiring the weight and showing it on the display.
- After loading of the goods, the user presses the REN (Rate Entry) button. The DWM enters into rate accepting mode and stays (Fig. 10) there until the user has finished all the 5-digits for rate. The digits appear on display as they are entered.
- At the end of 5-digits entry (000.00 – 999.99) for the rate, the DWM computes cost and shows it on the display. The display is updated once in 2-sec.

1.5 Things to be Done to Realize the DWM

Besides the mechanical aspects of the DWM, there are many hardware modules and software subroutines (SUR) that we need to design, develop and test. These are:

- Setup using MicroTalk-8051 Learning System (Fig. 12, 13),
- Collection of a goods holding frame including a load cell (Fig. 14),
- Calibrating the input device (Load holding frame, load cell, wiring and the ADC) to determine the gain and offset (Section-1.6),
- Create software SUR and download it into the RAM space of MicroTalk-8051 Trainer to acquire known weights (say, 1kg, 3kg and etc.) and showing it at DP6-DP10 (Fig. 3) positions of the display unit up to 3-digit precision,
- Create software SUR to acquire product rate from keypad on interrupt basis and showing at DP11-DP15 positions of the display unit,
- Create software SUR to compute cost, rounding it to 2-digit precision and showing it at DP0-DP5 positions of the display unit,
- Create software SUR so that the DWM responds to the keypad command ZER0 and nullifies the tare weight or any other stray/idle weight thereby.
- Create software SUR to generate 2-sec TT (Time Tick) using internal Timer-0 of the MCU for recurrent refreshing/updating of the display.

1.6 Calibration of the Input Device

In the DWM, we consider the input device as the aggregation of ***(i)*** the pane that holds the goods (Fig. 14), ***(ii)*** the holding frame that supports the pane (Fig. 13), ***(iii)*** the load cell, ***(iv)*** the wiring between the load cell and the ADC, and ***(iv)*** the ADC. Calibration of the input device refers to finding its gain and offset against known weights W1 (say, 1kg) and W2 (say, 3kg) and then deriving equation (1) for the unknown weight (W) in terms of the ADC output. This is known as 2-point calibration. In calibration process, we essentially transform a 'real system' into an 'ideal system' by imposing gain and offset on the real response (Fig. 2).

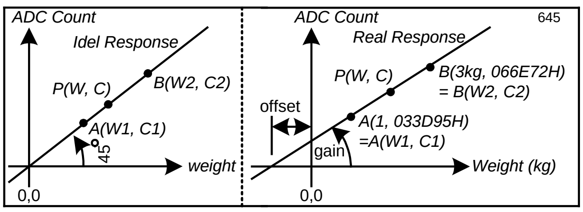


Fig. 2 Ideal and Real Responses of a System

The counts for points A(1kg, 033D95H) and B(3kg, 066E72H) are the averages of 10 readings each sampled in every 2-seconds. The recorded counts are:

- A (1kg, (033D85, 033DB3, 033E0D, 033D90, 033D40, 033DE6, 033DE1, 033DA3, 033DF3 and 033D65)),
- B(3kg, (066CB7, 066D0A, 066E2B, 066E63, 066F3A, 066FE9, 066F0C, 066E06, 066E2A and 066FCD)).

The response equation derived based on Fig. 12 is:

$$\frac{C2-C1}{W2-W1}=\frac{C-C1}{W-W1}$$

$$W-W1=\frac{C-C1}{C2-C1}*(W2-W1)$$

$$\Rightarrow W-W1=\frac{C}{0330DDH}*2-\frac{033D95H}{0330DDH}*2$$

$$\Rightarrow 10^4(W-W1)=\frac{C}{209117}*20000-\frac{212373}{209117}*20000$$

$$\Rightarrow 10^4(W-W1)=0.0956*C-20311.4046$$

$$\Rightarrow 10^8(W-W1)=956*C-203114046$$

$$\Rightarrow 10^8*W=956*C-203114046+100000000$$

$$\Rightarrow 10^8*W=03BCH*C-0625653EH\text{.................(1)}$$

Validity check of *Eqn. (1):*
10^8*W = 03BCH*066E72H-0625653EH (for 3kg)
$\Rightarrow 10^8$*W = 180471B8H – 0625653EH
$\Rightarrow 10^8$*W = 11DF0C7AH = 299830394
$\Rightarrow$W = 2.998 kg (correct to ± 2 gm)

Now, we may put forward the following algorithm to evaluate the expression of Eqn. (1) and display the cost on 7-segment display unit.

```
L1:    Initialize everything as needed

L2:    Check that conversion is complete and
       then acquire 24-bit (3-byte) output
       count (C) of the ADC. Save into registers:
       <R4 R3 R2>=066E72H for 3kg.
       LCALL  ACQADC

L3:    Compute 03BCH * C. Save the 40-bit (5-byte)
       result into:
       <R5 R4 R3 R2 R1> = 180471B8H.
       LCALL  MULT

L4:    Perform subtraction operation of:
       <R5 R4 R3 R2 R1> - 000625653EH and
       save the result in:
       <R5 R4 R3 R2 R1> = 11DF0C7AH.
       LCALL  SUBB

L5:    Convert 40-bit binary content of <R5
       R4 R3 R2 R1> into equivalent BCD using
       Horner Rule. Save result in:
       <R5 R4 R3 R2 R1> = <02 99 83 03 94>.

L6:    Divide the BCD result of L5 by
       100000000. Save result in:
       <R5 R4 R3 R2 R1> = <02. 99 83 03 94>.
       Keep weight in 3-digit precision.
       As a result, we have weight = 02.9980.

       There is no need to perform actual division
       by MCU instructions. Division process will
       be automatically done when we place the
       decimal point to the left of the 8th digit
       from right at the time of showing the weight
       on 7-segment display unit.

L8:    Show weight on 7-segment display
       devices with decimal point. We have:
       <DP6 DP7 DP8 DP9 DP10> = 0 2. 9 9 8.
       LCALL  CCWTX7SDD
```

2 HARDWARE BLOCK DIAGRAM OF 89S52 BASED DIGITAL WEIGHING MACHINE

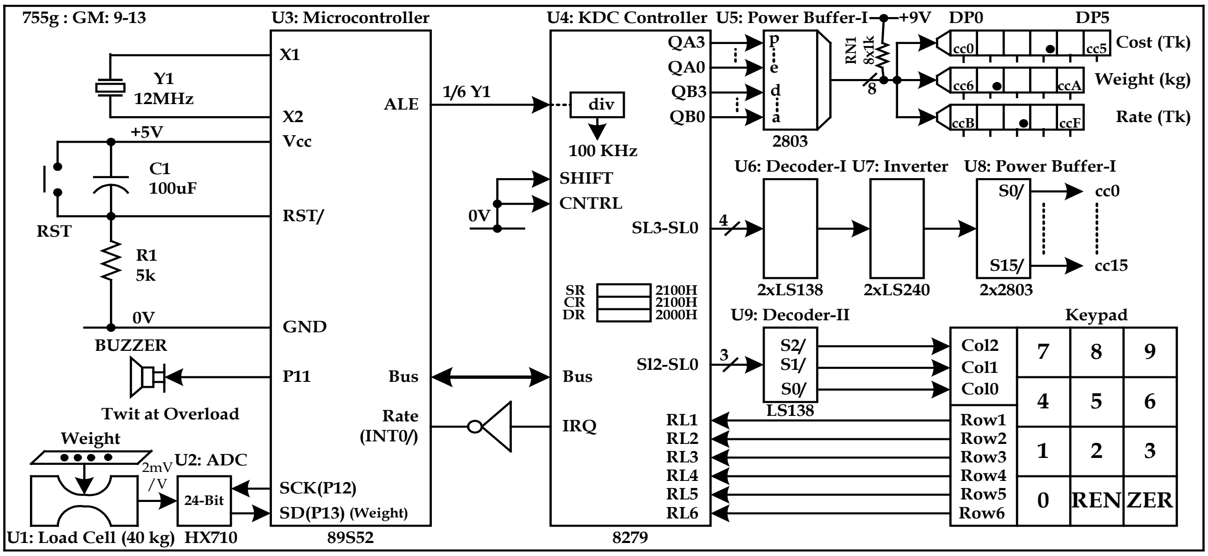


Fig. 3 Hardware Block Diagram of 89S52 Based Digital Weighing Machine

2.1 General Description

The load cell senses the weight and produces proportionate analog voltage, which is digitized by the ADC U2. The MCU acquires ADC's output, calibrates it, extracts the original BCD weight (BCDWT), converts it into CCWT (cc-coded weight) and shows at DP0-DP5 positions of the display. Product rate is entered from the keypad one digit at a time. The MCU collects the digit stream, consults memory-based lookup tables, extracts the original BCD rate (BCDRT), converts into CCRT and shows at DP12-DP15 positions. The MCU converts the BCDWT and BCDRT into BINWT (Binary Weight) and BINRT (Binary Rate) respectively. The BINWT and the BINRT are multiplied together to obtain BINCOST, converts it into BCDCOST and then into CCCOST, which in turn is shown at DP0-DP5 positions. At overload, the MCU makes twits on the buzzer and blanks the display unit.

2,2 U1: Load Cell

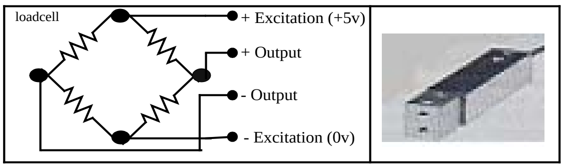


Fig. 4 Load Cell and its Electrical Equivalent Circuit

A load cell [1] is made of alloy aluminum with an embedded bridge network (Fig. 4) of which only two arms are subjected to load. Any unbalance created within the bridge is proportion to the applied load. A load cell is designed to have a nominal sensitivity of 2±0.1 mV/V of excitation at rated load. A 40kg load cell with excitation voltage of +5V will produce an output signal of 10mV.

2.3 U2: ADC

To digitize such a very low voltage signal of the load cell, we need a very precision and high gain ADC like HX710 [2]. HX710 is a 24-bit serial ADC with internal fixed gain of 128. This is a customized chip and has been specially designed for simple interfacing/wiring (Fig. 5) with weighing scale and the host microcontroller.

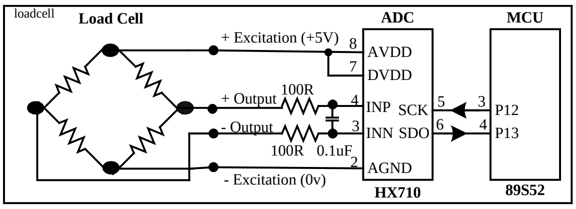


Fig. 5 Interfacing/Wiring of HX710 ADC

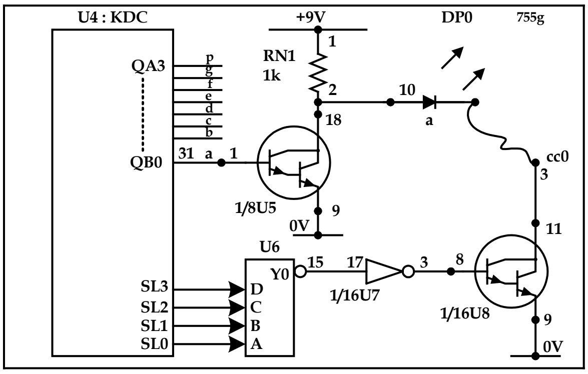
The ADC is automatically initialized during power up by the internal POR (power-on reset) circuitry. As long as the ADC conversion process remains active, the SDO (serial data out) line holds LH-state. At the end of conversion, the SDO line becomes LL and the ADC is ready to dump 24-bit serial data via SDO-pin with MS-bit first in response to twenty four SCK (serial clock). At the injection of 25th SCK pulse, the SOD-pin assumes LH-state. The ADC makes 10 conversions in 1-sec time. The following 8051 assembly codes accumulate the 24-bit data into registers <R4 R3 R2>.

```
;------------------------------------
ACQADC:;--weight data from ADC and save in <R4–R2>
       CLR P  1.2    ;SCK = LL
       SETB   P1.3   ;SOD is LL after conversion

CHK:   JB     P1.3, CHK ; conversion not complete
       MOV    R5, #18H  ; 24-bit data to read

READ:  ;----read data---------
       SETB   P1.2      ; SCK = LH
       NOP
       CLR    P1.2      ; SCK = LL
       MOV    C, P1.3   ; reads data bit from P1.3
       ;----------
```

```
MOV A, R2
RLC A
MOV R2, A
;--------
MOV A, R3
RLC    A
MOV    R3, A
;--------
MOV    A, R4
RLC    A
MOV    R4, A
;--------
DJNZ   R5, READ  ; <R4 R3 R2>
RET
;---------------------------
```

2.4 U5, U8: Power Buffers

Due to low power capability of the 8279 KDC (Keyboard/Display Controller), the brightness of the 7-segment display devices remain below acceptance level. This problem has been overcome by the introduction of the power buffers U5, U8. The following circuit of Fig. 6 explains the roles of the power buffers in delivering good amount of current through the segments of the display devices in order to increases the brightness of the display devices. Active signal at QB0-pin for segment-*a* of DP0 makes 1/6U5 OFF. At the same time, active signal at Y0-pin of U6 makes 1/16U8 ON. As a result current, whose value can be controlled by RN1 passes through segment-*a*.



Fig, 6 Role of Power Buffers to Enhance Light Intensity

2.5 Keypad

An active key is formed when the key is placed between row and column. The columns are excited either by walking 1's or walking 0's signals [3]. The row lines are accordingly terminated either to LL or LH through resistors. In the case of the keypad of Fig. 34, a 100 KHz walking 0's signals for the columns are provided by the SL2-SL0 scan lines of the 8279 and the 3-to-8 decoder (U9). The row lines are internally terminated to LH through pull up resistors. When a key is pressed down on the keypad, an 8-bit code (called Scan Code) is automatically generated within the electronics of the 8279 controller and is saved in a buffer called KBUF (keyboard buffer). The value of a scan code depends on the row and column across which the key is connected and strictly complies with the template format of Fig. 7. After detecting a pressed down key, the KDC controller puts LH at the 1st bit of its status register (SR). At he same time, the IRQ-pin also goes to LH-state, which interrupts the 89S52 MCU to inform that a key has been pressed down. When the user routine reads the scan code from the KBUF, the IRQ-pin comes down to the reset condition. The following Table-1 lists the scan codes for the keys of the keypad of Fig. 3.

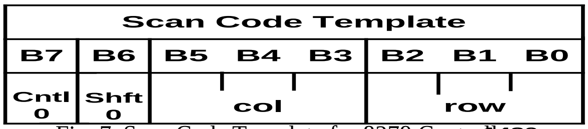


Fig. 7 Scan Code Template for 8279 Controller

Table-1 : Scan Codes		
Key	Key Label	Scan Code
K62	0	16H
K50	1	05H
K51	2	0DH
K52	3	15H
K40	4	04H
K41	5	0CH
K42	6	14H
K30	7	03H
K31	8	0BH
K32	9	13H
K61	REN	0EH
K22	ZERO	12H

2.6 50-mS Time Tick (TT) Generation

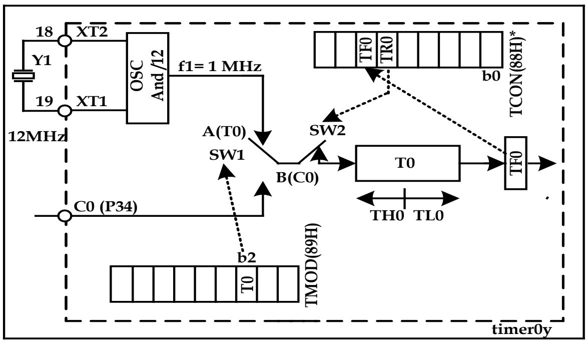


Fig. 8 2-sec Time Tick Generation Using Timer-0

Timer-0 (T0) of the MCU is configured to work in manual reload mode, where it starts counting up from a preset value and after counting exactly 50000 pulses from the internal clock source of 1MHz, it rolls over. The elapse time is 50mS and it has been defined as one Time Tick (TT). The 2-sec refreshing time for the display of the DWM has been derived from this TT. At the time of roll-over, the TF0-bit of the MCU assumes LH-state, which the MCU polls to ascertain that the 50mS time has elapsed. The following pseudo codes checks 2-sec time delay.

```
L1:    Load C350H as preset value into Timer-0
       Start Timer-0
L2:    if (TF0 != LH)
           Goto L2
L3:    CLR TF0          ; 50mS has elapsed
       Load T0 by c350H
       Increment  Counter
        If (Counter !=40)
           Goto L2      ; 2-sec has not gone
L4:    Refresh display by new data
```

3. CONTROL PROGRAM OF DWM

A Control Program (CP) for a microcontroller based system usually resides in the flash memory. Fig. 9 depicts the CP for the 89S52 based DWM

3.1 Flow Chart for Main Line Program

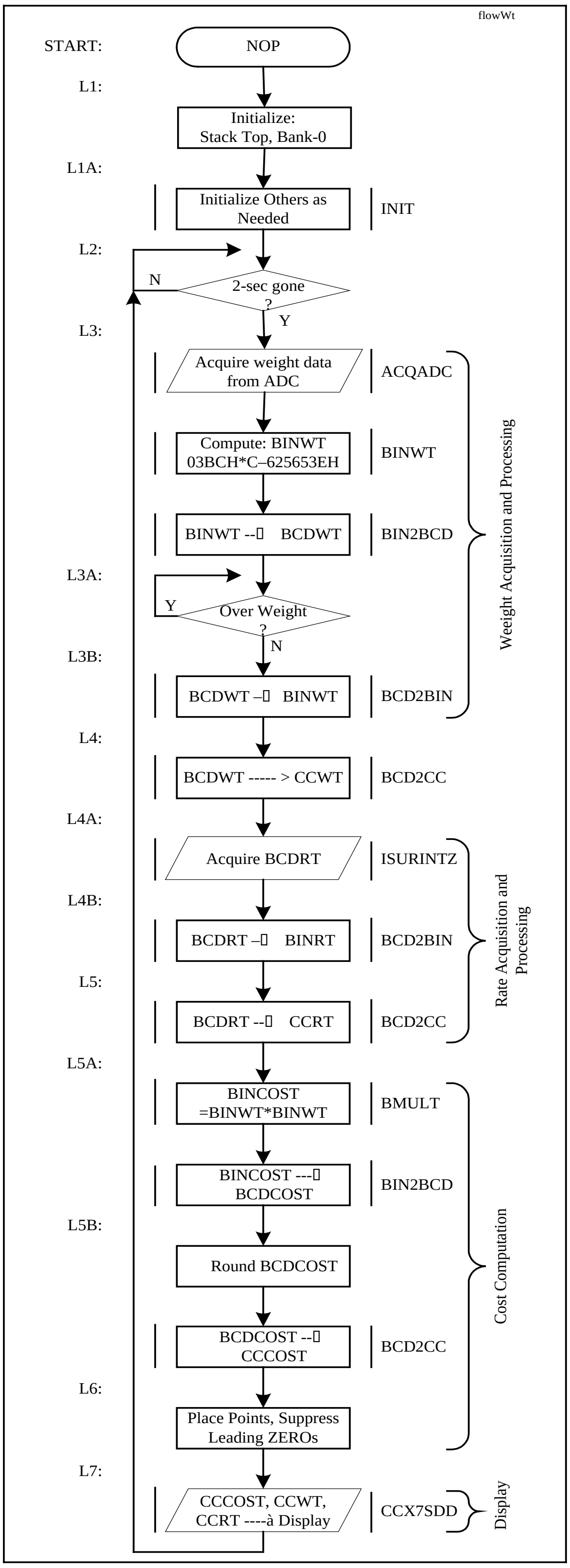


Fig. 9 Control Program Flow Chart for DWM

3.2 Rate Acquisition by Interrupt

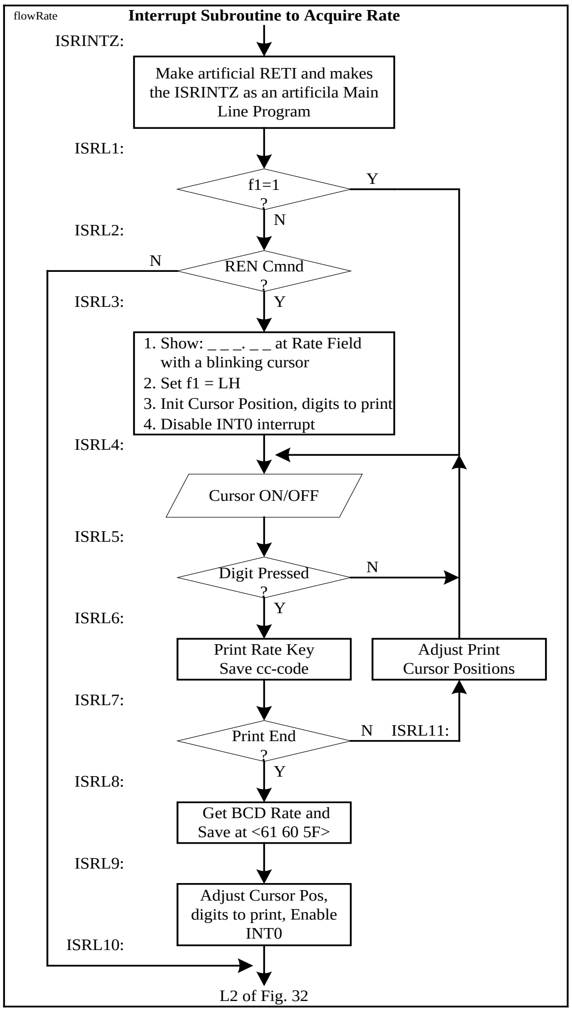


Fig. 10 Flow Chart for Rate Acquisition and Processing

The product rate that can be entered from the keypad has the range 000.00 to 999.99. Pressing the REN key interrupts the MCU and brings the DWM into rate accepting mode. The rate field of the display appears as _ _ _. _ _. After that the interrupt logic is reset by the execution of an artificial RETI instruction. As the digits for the rate (say Tk 456.75) are entered from the keypad, they arrive one after another on the rate field of the display as 4 _ _. _ _, 45 _. _ _, 456._ _, 456.7 _ and 456.75.

When a digit is pressed, the MCU collects its scan code, converts it into cc-code consulting a lookup table and shows the digit on the rate field of the display. The MCU saves the cc-code in a data buffer (5C, 5B, 5A, 59, 58, Fig. 11). At the end of 5-digit entry of the rate, the MCU recovers the original BCD rate from the stored cc-codes by consulting a lookup table. The BCDRT is converted to BINRT and then it is multiplied with BINWT to get the Cost. In Fig. 10, 11, we have presented flow chart and data structure for the rate acquisition and processing chain.

3.3 Data Structure

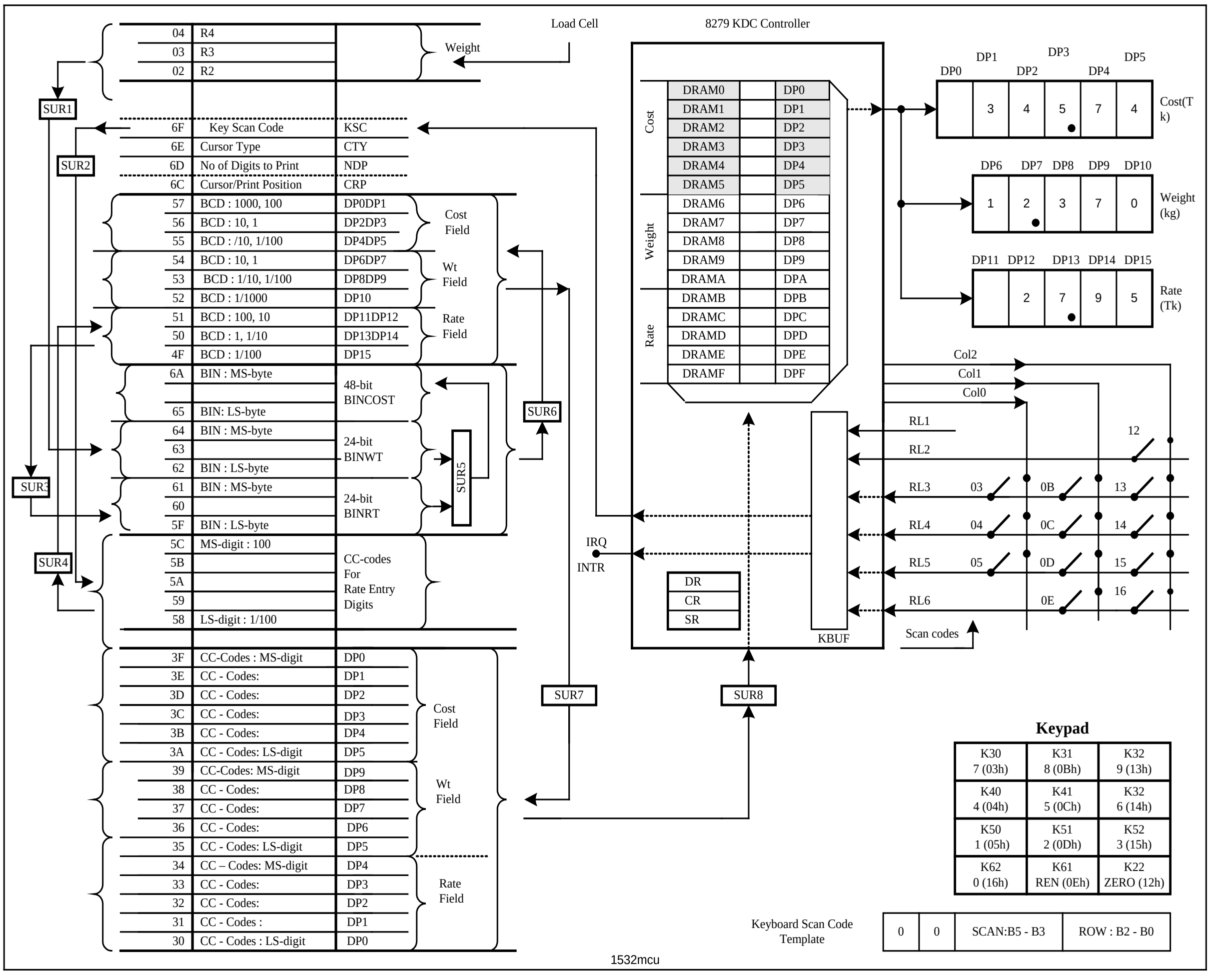


Fig. 11 Data Structure-II for the DWM

4. DEVELOPING DWM USING MicroTalk-8051

Fig. 12 Pictorial View of MicroTalk-8051 Trainer [4]

Fig. 13 Pictorial View of Mechanical to Hold Pane

Fig. 14 Pictorial View of Pane with a Weight

Fig. 15 Pictorial View of to be Developed Logic Board

5. CONCLUSION

DWMs are designed by the business houses and do not release the technical information for academic purposes. This paper has fulfilled this gap by providing experimentally verified codes, flow charts and methodologies, which the interested readers may exercise. ATmega32 based design may yield very simplified design by deleting the components U4-U9.

REFERENCES

[1] www.nbeastbright.com

[2] www.aviaic.com

[3] Mostafa, Golam: MicroTalk-8086 User Technical Reference Manual, 2004.

[4] MicroTalk-8051, User Technical Ref. Manual, 2006.

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Blended Learning Approach for Engineering Education –An Improvement Phase of Traditional Learning

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Abstract—**Engineering education is so complex to understand when teaching by a teacher face-to-face (F2F) with the students in our academic education system. This traditional teaching approach cannot make interesting the lecture at classroom and student cannot get attention to the class due to the absence of interactive presentation.To overcome the problemof the traditional teaching of engineering education we demonstrate architecture of blended learning (BL) that helps learners to bring their experiences and ideas to the intellectual conversion, the understanding of the other participants is enriched, resulting in active learning.This paper describes various event-based activities, including F2F classrooms, live eLearning, and self-paced learning.We have proposed a systematic way of BL approach with the existing infrastructure of any institutions.Organizations must use BL approaches in their strategies to get the right content in the right format to the right people at the right time. BLcombines multiple delivery media that are designed to complement each other and promote learning and application-learned behavior especially in engineering education. A prototype system is developed and tested using different learning scenarios. The system has also been tested by a group of students.**

Keywords— Blended learning; face-to-face interaction; Traditional learning;Engineering education;Self-regulated learning.

I. INTRODUCTION

Blended learning approach for engineering education system enriched the learners learning capacity than traditional learning system. This paper suggest a cost-effective architecture to transform existing learning spaces into effective spaces that enable better learning and collaboration, given the resource limits of a university setting.Description about the BL isdescribed in the Section-II. Section-III and Section-V respectively describeabout the frame work of BL and Case studies. There are two methodology has provided to solve the percentage of BL with respect toengineering subjects. Self-Regulated Learning (SRL) and Blended Learning for Engineering (BLE) discussed in Section–VI. For doing a survey on the university students of engineering discipline we have developed a website in Section-VII. Data collection and experimental result described in Section-VIII and Section-IX respectively. We have proposed a architecture of BL approach which is better and successful system of teaching style than traditional teaching approach described in Section-XI.

A cost-effective infrastructure needsfor building ubiquitous collaborative learning spaces. It uses techniques from the Semantic Web and ubiquitous computing to build a learner-centric service-based architecture to transform existing traditional learning spaces (e.g., classrooms, computer labs, meeting rooms, and hallways) into intelligent ambient learning environments. This is achieved by blending a number of inexpensive technologies which are optimally configured to provide services that can perceive a learners' location and schedule, identify current learning activity, recommend learning resources, and enable effective real-timecollaboration and resource sharing between learners and their instructors [1]. This paper suggests a cost-effectivearchitecture to transform existing learning spaces into effective spaces that enable better learning and collaboration, given the resource limits of a university setting. Interaction analysis can help understand the practice and development of Self-Regulated Learning (SRL) in Virtual Learning Communities (VLCs). To this end, a set of SRL indicators is proposed to spot clues of self-regulated events within students' messages[2].Practical experiences acquired in design, realization and implementation of interactive e-learning project located on the educational portal for students called "eLearn central". This portal is permanently being used in the distance and blended learning at Slovak University of Technology University, in the popularization of Science and Technology between kids and young people and for team work in everyday business life [3]. Over the past decade, online learning has become an increasingly popular option among post-secondary students. Yet the higher education community still regards fully online courses with some ambivalence, perhaps due to the mixed results of a large (if not necessarily rigorous) body of research literature. On the one hand, research suggests that students who complete online courseslearn as much as those in face-to-face instruction earn equivalent grades, and are equally satisfied [4]. On the other hand, online students' are less likely to complete their courses [5].Online learning adaptation which contains five aspects: online learning environment, online learning motivation, online learning mode, online learning ability, online learning efficacy and achievement. The research takes the example of Guangxi University undergraduates to survey online learning adaptation. The results show that the total level of college students' online learning adaptation was relatively low [6]. In many cases, student learning outcomes in online courses are superior to those in traditional face-to-face courses [7].To address the issue of quality assurance and online teaching and learning the authors are looking at a two phase process, the first of which is the development often initial audit tool examining online technical aspects followed by the collaborative development of a peer review process centered on pedagogical issues [8].

II. WHAT IS BLENDED LEARNING

Blended learning is a formal education program in which a student learns at least in part through online delivery of content and instruction with some element of student control over time, place, path, and/or pace. While still attending a "brick-and-mortar" school structure, face-to-face classroom methods is combined with computer-mediated activities.Proponents of blending learning cite the opportunity for data collection and customization of instruction and assessment as two major benefits of this approach. Schoolswith blended learning models may also choose to reallocate resources to boost student achievement outcomes [9]. Blended learning as a formal education program in which a student learns at least in part through online learning, with some element of student control over time, place, path, and/or pace; at least in part in a supervised brick-and-mortar location away from home; and the modalities along each student's learning path within a course or subject are connected to provide an integrated learning experience [10].

III. FRAMEWORK OF BLENDED LEARNING

F2F classrooms, live eLearning, and self-paced learning are mixed of traditional instructor-led training; synchronous online conferencing or training, asynchronous self-paced study, and structured on-the-job training from an experienced worker or mentor. This BL system follows the following purposes.

Table 1. Categories of Blended learning Approach

Synchronous physical formats	Instructor-led Classrooms & Lectures Hands-on Labs & Workshops Field Trips
Synchronous online formats (live e-learning)	Online Meetings Virtual Classrooms Web Seminars and Broadcasts Coaching Instant Messaging Conference Calls
Self-paced, asynchronous formats	Documents & Web Pages Web/Computer Based Training Modules Assessments/Tests & Surveys Simulations Job Aids & Electronic Performance Support Systems (EPSS) Recorded Live Events Online Learning Communities and Discussion Forums Distributed and Mobile Learning

IV. BACKGROUND AND RELATED WORK

As mobile and wireless technology rapidly advances, newphysical and virtual learning spaces are changing the waylearner's access and share resources, acquire knowledge,and interact and collaborate with each other. In thesemodern ubiquitous learning spaces, technology can movebeyond the relatively predictable wired classroom computers and dissonant presentation systems. Learning technology expands to include diverse embedded sensors,wireless instructional devices such as handheld computers, and a variety of interconnected technologies. These platform advancements pave the way for context awareness, ubiquitous computing, and semantic web technologies to create innovative learning interactions. These enhancements allow learning to expand beyond the classroom to labs, field-trip locations, meeting rooms, and even hallways and study areas. Thus, offering various cooperative learning opportunities, enabling a higher level of reasoning, a deeper level of understanding, a greater motivation to learn, and greater social competencies [11]. With allthese concerns, designing ubiquitous collaborative learningspaces becomes a complex and challenging task thatinvolves many computational and learning paradigms.This paper proposes a solution to some of these challenges,in particular, those related to context perception andmanagement of learner/activity

mobility, social andintellectual interactions, and seamless knowledge disseminationand sharing. Before addressing these challenges, wefirst survey some of the ubiquitous learning systemsreported in the literature.There are many existing technologies that have thepotential to be developed into powerful learning enhancementtools [12, 13, 14, 15, 16, 17]. The uClassroom [18]project is a more modern ubiquitous computing environment that is designed for school applications, created at Nagoya University of Japan. It uses context to provide optimized educational information via a web interface. While uClassroom mentions location data as part of its context awareness, it is not intended to provide any potential system for attaining this location information. Additionally, it does not involve the communication between individual systems, limiting potential applications in collaborative and peer-to-peer situations.
KERIS, a leading e-learning organization in Korea, hasdeveloped a ubiquitous smart classroom environmentcalled uClass [19].

Most of the references that we mentioned in this paper describe about the e-learning based education system where they have told about the learning process through only web services. Reference [1] mentioned only about the learning spaces. They described about the spaces of classroom, computers lab, meeting room and hallways and existing research going on blended learning through mobile based services. But these papers don't mention that how the blended learning approach can be applied in the class-room of engineering education?Our proposed idea has mentioned the blended learning approach that combines both F2F interaction and online services through our website. Since it is regulated by teacher so its importance not negligible and also analyzed the percentage of blended learning components that helps to make interesting lecture.

In some universities of Bangladesh, teachers provide their lecture at the beginning of the semester but the teaching method doesn't differ from the traditional techniques. We are going to propose a new form of engineering education system that will make it more effective and realistic.

V. CASE STUDIES

To analysis the different engineering courses on the basis of BL we have divided our academic educational system into three categories, such as Arts, Science and Commerce. Under these categories the different subjects (we have analyzed only five subjects) are compared with BL approach by the help of teachers and students. This comparison was done on the basis of how a teacher makes lecture interactive with the help of BL components and how students understand a lecture easily. Here MR represents Maximum required and LRrepresents less required.

Table 2.Necessity of Blended Learning in Different Courses Under Three Disciplines

Discipline	Subject	F2F	e-learning	Presentation	Audio	Video	White-Board
Arts (Humanities)	Political Science	MR	LR				
	Economics	MR		LR			MR
	English	MR			LR		MR
	History	MR	MR				
	Geography	LR		MR		MR	
Science	Physics	MR	LR	MR			LR
	Mathematics	MR		LR			MR
	Chemistry	MR	LR	MR			MR
	Biology	LR		MR		MR	LR
	Computer	MR	MR	MR	LR	MR	LR
Business	Finance	MR		LR			MR
	Accounting	MR		LR			MR
	Management	LR	MR	MR			
	Marketing	LR		MR	LR	LR	
	Banking	MR	LR	MR			

Since traditional learning approach cannot make lecture interactive especially in the discipline of engineering, it is quite impossible to accurately understand the engineering subjects. So we have performed a survey on the ICT discipline of engineering categories with the interviewing of teachers and students. In this table we find that every subject of ICT discipline mostly required BL components to present and understand a subject.

Table 3. Necessity of Blended Learning in Different Subjects of ICT Disciplines

Subject	F2F	e-learning	Presentation	Audio	Video	White-Board
Algorithm	LR	MR	MR		MR	LR
Microprocessor	MR	LR	MR		MR	LR
Operating System	LR	MR	MR		LR	
Data Structure	LR	LR	MR	MR	MR	LR
Database	LR	MR	MR		LR	
Distributed System	LR	MR	MR		MR	
Graphics	LR	MR	MR			LR
Artificial Intelligence	MR	MR	LR	LR	MR	
Computer Architecture		LR	MR		MR	
Networking	LR	LR	MR		MR	LR
Telecommunication	LR	MR	MR		LR	LR

Above table were the engineering categories of ICT discipline. There without BL, subject lecture cannot make interactive to the classroom. From Table-2 and Table-3 we have concluded that the engineering discipline of different departments needs BL approach to presents the class lectures effectively. So traditional learning cannot ensure the possibility of learning capability successfully that already done by our BL approach.

Table 4. Comparison of Different Engineering Discipline on the Basis of Blended Learning

Engineering Discipline	F2F	e-learning	Presentation	Audio	Video	White-Board
EEE	MR	MR	LR			MR
CSE	LR	MR	MR	LR	MR	LR
ECE	LR	MR	MR	LR	MR	LR
MEE	MR	MR	LR			LR
Civil	MR	LR	MR			MR
Textile	MR	MR	LR		LR	LR
Architecture	MR		MR	LR	MR	

VI. METHODOLOGY

A. *Self-Regulated Learning (SRL):*

SRL helps to the students to improve in engineering education which is cumulativewith blended learning. It also guides the student's abilities to plan, monitor, and evaluate their own learning process; these can be investigated by spotting the learner's active contribution to: choosing learning objectives and contents; working out or adapting learning strategies; suitably configuring the learning environment; evaluating learning results by comparing one's outcomes with the outcomes of peers and with models possibly provided.

B. *Blended Learning for Engineering (BLE):*

$$\mathrm{BL} = \frac{(\mathrm{F2F} + \mathrm{EL} + \mathrm{PPT} + \mathrm{AD} + \mathrm{VD} + \mathrm{WB})}{\text{Total No. BL Components}}\, 100\%$$

$$f(BL) = \frac{\sum_{i=0}^{n}(BLC)}{N}\, 100\%$$

F2F→ face-to-face, EL→ e-Learning,
PPT→Presentation, AD→ Audio, VD→ Video,
WB→Whiteboard, BL→ Blended Learning

We have identified anequation to find out the required percentages of blended learning approach on the different discipline of our education system. From the above discussion we find that the following tables percentage ratio of blended learning. Whereengineering disciplines are required 78% our approach of Blended learning.

Table 5. Percentages of BL approach

	Applied Science (%)	Arts (%)	Business (%)	Engineering (%)
Blended Learning (BL)	65%	40%	55%	78%

VII. WEBSITE DEVELOPMENT

We have developed a programming tutorial site in Banglausing ASP.NET4.5 MVC4, Entity Framework 5.0 and Razor view engine anduploaded it on the internet [10].We have developed it to give tutorials on different programming languages so that Bangladeshi students can easily understand the programming problems, get ideas about the problems, generate ideas and implement it in the program [20].

Figure 1. Bangla programming website

VIII. DATA COLLECTION BASED ON LEARNIG PROCESSANDEXPERIENCE

We did a survey on the 2nd and 4th semester students of PSTU, CSE on the basis of BL. We provided programming related tutorials to both 2nd and 4th semester students. Video and text-image-animation based tutorials were provided to the students that distributed into two ways. We divided the students into two categories. i) The students whose are going to learn JAVA programming and ii) The students whose already learned JAVA programming.

Firstly video tutorials then text-image-animation were given to the 2nd semester and secondlytext-image-animation then video tutorials were given to the 4th semester students. Both levels of student were prefer to video than text-image-animation based tutorial for understanding programming like Java, C/C++. This process was completed by our developed website and also given opportunity to collaboratively learn through social site or group discussion before the class lecture day. They practiced the lecture topic and come to the class room where teacher and students discussed F2F with one another. As a result, the students could be able to giving response quickly and they had understood the lecture topics easily. So idea's identified by F2F interaction of teachers and students in the class room.Problems they faced and solved the problems. Then we tried to find new ideas related to the problems and what can be the solution of those problems. After solving the problems we collected an online survey [21]. Answers of the questionnaire from the respondents are represented by different chart in the following sub-sections.In our survey we selected 100 participants. From the participants both of 2nd Semester and 4th Semester were 50%. They were divided into two groups and attended in two categories. We took the survey in two different times from two groups.

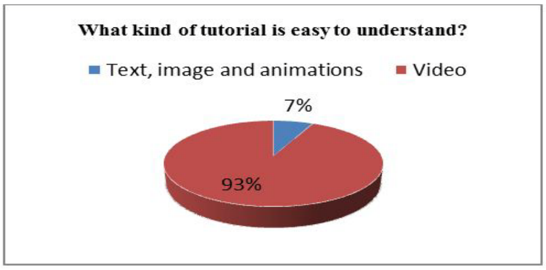

Figure 3. Pie chart of answer of the question

From the above chart we have found that participants can easily understand any problems from video tutorial than website textual format. Only 7% like to see text, image and animation related tutorials.

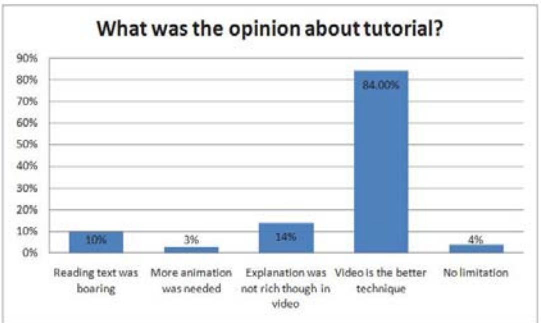

Figure 4.Column chart of answer of the question

We divided our total sample into 5 categories based on tutorial pattern and they have given their opinion where- 10% reading text don't like, 3% need to more animation, 14% explanation was not rich, video is better technique and 4% no limitations.

IX. EXPERIMENTAL RESULT

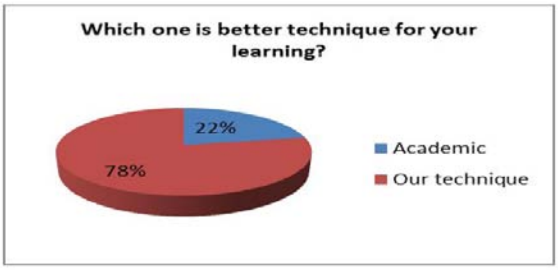

Figure 6.Pie chart of answer of the question

Hopefully, 78% students supported our teaching technique and only 22% students told that the academic learning technique is better. So we have found a positive sign for our research.

X. PROPOSED IDEA

Face to face teaching system in the class room with only white board and marker cannot make interactive lecture first time of any subject. In engineering discipline's most of the studentscannot get more attention to the class lecture because of traditional learning system. To improve the teaching and learning of traditional learning system we have proposed that the teachers' will upload his/her lecture/tutorial with video, audio, presentation or other related resource of a lecture in the university website before the class day. The students will download these resources and see it first then read the text lectures/tutorials;they also share this lecture with group through social media or other process so that they will be more able to understand about topics. After acquisition knowledge on that topic, students will come to the class room. On the next day the teacher and students can discuss with the topic and then the students can ask questions, provide their ideas easily. Therefore the class can be more effective than the traditional teaching system. This blended learning (BL) approach is indispensable now in the engineering institutions for better understanding in engineering education.

XI. ARCHITECTURE AND SYSTEM DESIGN OF BLEANDED LEARNING

To alleviate the traditional teaching problem of engineering subjects we demonstrate anarchitecture of blended learning that helps learners to bring their experiences and ideas to the intellectual conversion, the understanding of the other participants is enriched, resulting in active learning. According to our developed system, at first student will download lecture from the website before the class day and after group studying in any location through different online/offline process they will come to the class room on the next day. Teacher and students will discuss about that topic of lecture in the class room through multimedia presentation. They will share their gathered knowledge, discuss with the problems, identify the solutions of the problems and then implement the solution. Consequently they will able to solve the problems easily and getting attention to the class lecture. Then their idea and implementation will upload to the university server. This process is the continuous process.

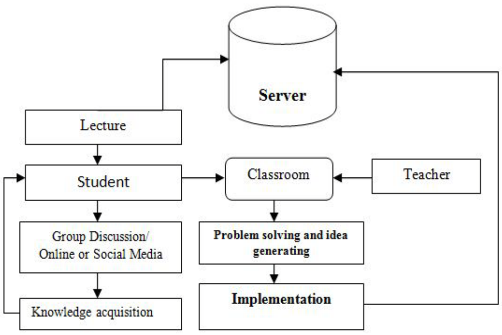

Figure 7.System design of BL for Engineering Education

XII. CONCLUSION AND FUTURE WORK

Traditional learning should be avoided to better understanding in engineering education. To develop in engineering education need the blended learning approach. Students and teachers can collaboratively analyze the problem definition. Technology enhances learning capabilities of learners by our proposed work. Blended learning approach helps to understand engineering subject at easier fashion.Increase the knowledge of education through social interaction. We took only 100 students as our sample and our efficiency of BL approach is 78%. Our approach can be easily configured to any institutions because it can be applied with existing infrastructure of university class room. Because most of the universities already have multimedia projectors, internet, internal server, e-library and sound system based class room. In future we want an integrated BL system for all type of education system.

REFERENCES

[1] K. Scott, Senior Member, IEEE, and R. Benlamri, Senior Member IEEE. "*Context-Aware Services for Smart Learning Spaces*".

[2] Giuliana Dettori and Donatella Persico. "*Detecting Self-Regulated Learning in Online Communities by Means of Interaction Analysis*"

[3] Lubica Stuchlikova, Jana Benkovska. "*eLearn central- the journey to e-learning*".

[4] Jahng, N., Krug, D., & Zhang, Z. (2007). "*Student achievement in the online distance education compared to face-to-face education. European Journal of Open, Distance and E-Learning*".

[5] Beatty-Guenter, P. (2002, May). "*Why distance education? Mimetic forces in institutions of adult education*." Paper presented at the 21st Annual National Conference of the Canadian Association for the Study of Adult Education, Toronto, Ontario.**(Conference)**

[6] ShuanglanLuo, Dept. of Educ. Technol., Guangxi Normal Univ., Guilin, China ; Xueqin Huang," *A survey research on the online learning adaptation of the college students*"

[7] Shanna Smith Jaggars and Thomas Bailey,"*Effectiveness of Fully Online Courses for College Students: Response to a Department of Education Meta-Analysis*".

[8] Robert M. Corderoy, Ray Staceand R. Pennell,"*Quality assurance and online teaching and learning: first steps*".

[9] (2013, July) http://en.wikipedia.org/wiki/Blended_learning**(Link)**

[10] (2013, July)http://www.christenseninstitute.org/blended-learning-3/**(Link)**

[11] J. Barbosa, R. Hahn, S. Rabello, and D. Barbosa, "*LOCAL: AModel Geared Towards Ubiquitous Learning.*" ACM SIGCSEBull., vol. 40, no. 1, pp. 432-436, 2008.

[12] Y. Rogers, S. Price, C. Randell, D.S. Fraser, M. Weal, and G.Fitzpatrick, "*Ubi-Learning Integrates Indoor and Outdoor Experience,*"Comm. ACM, vol. 48, no. 1, pp. 55-59, 2005.

[13] Y.E. Shih and D. Mills, "*Setting the New Standard with MobileComputing in Online Learning*" Int'l Rev. of Research in Open andDistance Learning, vol. 8, no. 2, pp. 1-16, 2007.

[14] T. Wu, T. Yang, G. Hwang, and H. Chu, "*Conducting SituatedLearning in a Context-Aware Ubiquitous Learning Environment*"Proc. Fifth IEEE Int'l Conf. Wireless, Mobile, and UbiquitousTechnology in Education, pp. 82-86, Mar. 2008.226 Ieee Transactions On Learning Technologies, Vol. 3, No. 3, July-September 2010

[15] H. Ogata, "*Computer Supported Ubiquitous Learning: AugmentingLearning Experiences in the Real World*" Proc. IEEE Int'l Conf.Wireless, Mobile, and Ubiquitous Technology in Education, pp. 3-10,Mar. 2008.**(Conference)**

[16] T. Chen, C. Chang, J. Lin, and H. Yu, "*Context-Aware Writing inUbiquitous Learning Environments*" Proc. Fifth IEEE Int'l Conf.Wireless, Mobile, and Ubiquitous Technology in Education, pp. 67-73,Mar. 2008. **(Conference)**

[17] Chen, C. Chang, J. Lin, and H. Yu, "*Context-Aware Writing inUbiquitous Learning Environments*" Proc. Fifth IEEE Int'l Conf.Wireless, Mobile, and Ubiquitous Technology in Education, pp. 67-73,Mar. 2008.**(Conference)**

[18] S. Kajita, K. Mase, S. Jang, M. Ueda, Z. Yu, and N. Lin,"*uClassroom: Expanding Awareness in Classroom to UbiquitousTeaching and Learning Using Eclipse RCP*" Proc. EclipseCon, Mar.2007.**(Conference)**

[19] K. Kim, "*Ubiquitous Learning Supporting System for FutureClassroom in Korea*" Proc. Soc. for Information Technology andTeacher Education Int'l Conf., K. McFerrin et al., eds., pp. 2648-2657,Mar. 2008.**(Conference)**

[20] (2013, June)http://programmers.somee.com/**(Link)**

[21] (2013, July)https://docs.google.com/forms/d/1y4SPNEtavZQZeI9ZPFcH4brzXvtofky8We_JHhjO2A/viewform**(Link)**

An Efficient Approach to Save Cluster Head Energy of Sensor Network

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***Abstract*— Wireless sensor network is an emerging technology in wireless network. Improving the lifetime of sensor node is the major issue for designing the sensor network. For these reason the main concern of researchers is how to utilize the medium in a power effective manner. With advance of these, various MAC protocols are introduced and lots of recent works are going on. But, most of these MAC fail to maintain the effective tradeoff between power consumption and latency. LEACH protocol is one of them which are based on clustering techniques. In this paper, we propose a method for intra cluster communication techniques that decrease power consume by cluster head node also latency and compare with traditional LEACH protocol.**

Keywords: Wireless Sensor Network; MAC protocol; Enhancing Lifetime; Cluster Head.

I. INTRODUCTION

Wireless Sensor Network is the most promising technology in the last decade during its robust application like environment monitoring, robotic exploration, military application and many others. With this advent of the rapid application, some critical situations are come into front. Power consumption is one of the most critical situations. Because, when the power of nodes are going down it is too much impractical to replace the node or power up of these node. Researchers in this area should keep in mind the energy dissipation factors.

Wei Ye et al.[1], llker Demircol et al. [2] Identified some major sources of energy waste. Collision is first one. When a transmitted packet is corrupted it has to be discarded, and the follow-on retransmissions increase energy consumption. Collision increases latency as well. The second source is overhearing, meaning that a node picks up packets that are destined to other nodes. The third source is control packet overhead. Sending and receiving control packets consumes energy too, and less useful data packets can be transmitted. The last major source of inefficiency is idle-listening, i.e., listening to receive possible traffic that is not sent. This is especially true in many sensor network applications. If nothing is sensed, nodes are in idle mode for most of the time. However, in many MAC protocols such as IEEE 802.11[3] or CDMA nodes must listen to the channel to receive possible traffic. Many measurements have shown that idle listening consumes 50–100% of the energy required for receiving. Different types of protocols are used to avoid that energy wastage. Most of them are basically application specific. LEACH protocol is one of the impressing protocols which use clustering method for saving energy. But it increase delay. In this paper we propose an intra cluster communication method and average energy based cluster head selection technique for achieving better performance in context of both less energy consumption and less delay.

The remaining part of this paper is organized as follows: related work in Section II. In section III, the assumption and network architectures are discussed. Section IV describes proposed methodology, section V describe the experimental result. Finally, section VI concludes the paper.

II. RELATED WORK

We said earlier that the main concern of most MAC protocol in the sensor network field is to reduce the energy consumed due to idle listening, overhearing, collision etc. To eliminate these drawbacks many protocols are drawn. Some protocols pay attention to reduce idle listening, where some other's attention to minimize latency. Some of them are contention based, some others are schedule based. IEEE 802.15.4 protocol combines both contention based and schedule based elements to achieve significant commercial impact [4]. S-MAC and T-MAC [5] use duty cycling method to reduce idle listening. S-MAC use fixed duty cycling, however it gives poor result for varying traffic load. T-MAC use dynamic duty cycling method over S-MAC to adapt it in various traffic load situation. But, it bottleneck by early sleeping problem. Both these protocol used Network Allocation Vector (NAV) to escape overhearing but it put forward idle active situation. Synchronization of duty cycle [6] also introduced to maximize sensor network performance. PAMAS [7] initiate a busy tone technique where other neighboring nodes need not to wait a NAV amount of time. When it hears busy tone, it permits to go to sleep mode. DW-MAC [8] is a low-overhead scheduling algorithm that allows nodes to wake up on demand during the Sleep period. AS-MAC [9] works over DW-MAC, it use Adaptive

Scheduling (AS) period into the operational cycle within which the nodes' active duration is made adaptive to variable traffic load. TRAMA [10] is a traffic adaptive protocol design whose main concentration is to reducing latency by using random access and schedule access technique. For doing these Venkatesh Rajendranet.et al. use three other techniques, neighbor Protocol (NP), Schedule Exchange Protocol (SEP) & Adaptive Election Algorithm (AEA). To perform all three techniques attractive CPU requirement must fulfill and it utilizes much memory compare to others.
Later LEACH [11], an application specific protocol architecture developed. Which offer a cluster based self-adaptive and randomized technique to save power. In LEACH all nodes forming cluster and every cluster must have a cluster head node. These is responsible for collecting data from all non-cluster head node under its cluster using TDMA and then aggregate the data and send it to the base station (BS) using CDMA. Therefore, cluster head is more energy intensive than other node. So, if a node continuously remains as a cluster head node it decays its whole energy and after some round it falls. The LEACH incorporates randomized rotation of a high energy cluster head position such that it rotates among the sensors in order to avoid draining the battery of any one sensor in the network. In this procedure the power load is evenly distributed among the nodes.

III. PROBLEM STATEMENT

As we discuss in section I, we work over LEACH protocol and find out some problems. One is idle listening by Cluster Head node. By using LEACH intra cluster communication is based on TDMA. The cluster head announce the time schedule for each non-cluster head node under its cluster. The cluster head remains awake during the whole time slot. It is very much energy consuming in context of lower traffic. Another problem is latency in crucial situation. Consider a situation where nodes are isolated from one another at a good distance.

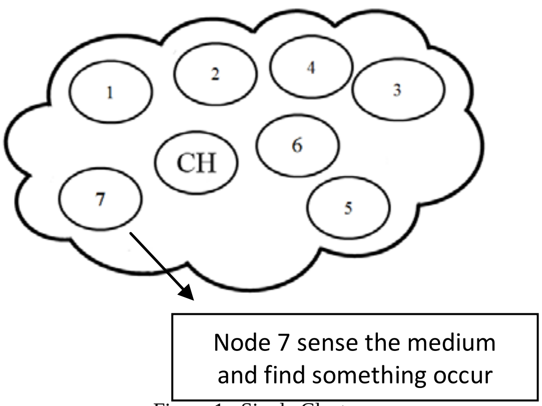

Figure 1. Single Cluster

In figure (1), when node 7 senses the medium it finds something occur and it wants to transmit data to the cluster head. Whereas, cluster head previously announced the schedule of non-cluster head node. On this schedule node 7 gets its slot at very late period say after 30 ms [figure (2)]. Though the other nodes who are getting the previous time slot have no data to send. Thus it increases latency.

0	5	10	15	20	25	30
1	2	3	4	5	6	7

Figure 2. Time framing in TDMA

IV. DESIGN OF PROPOSED MODEL

To fulfill the ultimate goal of this paper, we arrange this section into two parts. Where Enhancement of cluster head selection algorithm followed by CSMA based intra cluster communication techniques takes place. Enhancement of cluster head selection algorithm is the extension of LEACH's cluster head selection procedure. It can keep all nodes power into an average level and prevent sudden power down of nodes.

A. Enhancement of Existing cluster head selection algorithm

Assume that existing cluster head selection algorithm primarily select 10% cluster which we called Primary Selected Cluster Head (PSCH). Among them our proposed algorithm reduces 50% cluster head on the basis of average power. Since longevity is one of the main concern of LEACH protocol. So, we tried to keep equal power among all nodes. Enhancement of cluster head selection algorithm is given below.

Algorithm:

Step 1: Set a counter "CHCounter" for every sensor, which determine the number of times it elected as cluster head.

Step 2: Calculate "avg_energy" (average energy).
Calculate avg_energy of the sensor node among the Primary Selected Cluster Head using the following formula

$$\text{avg\_energy} = \sum_{i=0}^{n} \frac{E_i}{n} \qquad (1)$$

Here, n = number of Primary Selected Cluster Head (PSCH), E_i = Energy of each PSCH.

Step 3: Repeat step 4 and 5 until it reduces 50% of Primary selected cluster Head (PSCH).

Step 4: Pick a PSCH (Pick one after another. First, choose PSCH which have CHCounter = 0)

Step 5: Choose cluster head

a. If CHCounter = 0 and energy of the picked PSCH is greater than or equal of the avg_energy; select it as CH (Finally selected Cluster Head) and go to step 3

Otherwise if power of the picked PSCH is greater or equal of the avg_energy; select it as CH (Finally selected Cluster Head) and go to step 3

Step 6: Exit

Flowchart of the algorithm:

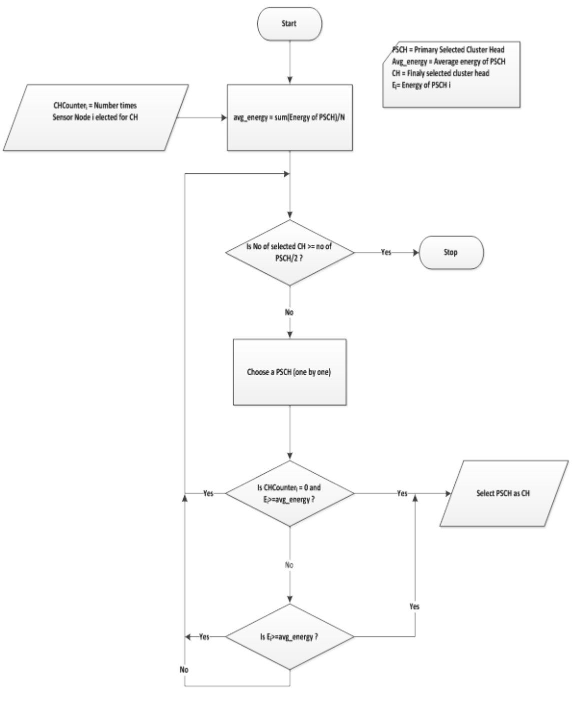

Figure 3. Flow chart (Enhancement of exiting cluster head selection algorithm)

B. *CSMA based Intra-cluster communication technique*

Typically LEACH use TDMA for intra-cluster communication technique for transmitting data from non cluster-head node to cluster head node. In section III, we discuss some problems by using TDMA. Now, here we initiate a CSMA based technique use preamble sampling. WiseMAC [12] and B-MAC [13] uses preamble sampling technique. WiseMAC, used dynamic preamble technique for downlink communication whereas B-MAC used fixed length preamble by using Clear channel Assessment (CCA). Like both of these MAC we sample the medium at fixed time length T_w. After T_w time the cluster head node awake and seek channel for data. If cluster-head found nothing it go to sleep mode. It senses the channel during its whole round. Now, it's a common question that what should be the size of preamble?

Size of preamble > Two * wakeup Time +Sleep Time (2)

When two or more nodes want to transmit at the same time it first sense the channel in figure (4), if it finds the channel is occupied by other node it takes a random amount of time and then again sense. These procedure repeats until found the channel idle. Thus this technique is well suited for low traffic and where these nodes are placed for long term monitoring e.g. forest fire.

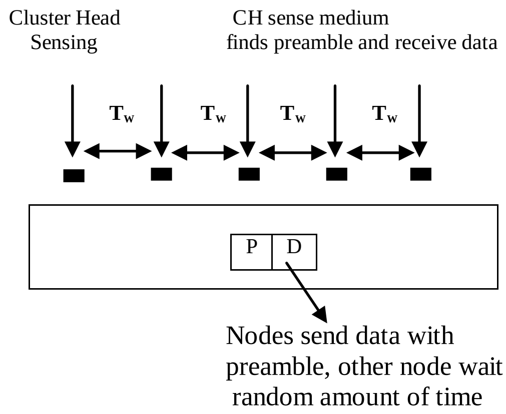

Figure 4. Model of our proposed method

V. MATHEMATICAL ANALYSIS

In this section we show some mathematical calculation and simulation. Here we show the comparative analysis how cluster head consume energy of typical method and our proposed method. Also show the delay analysis between those two. We use MATLAB for our simulation.

In this analysis we only show the energy consumption by cluster head node and delay of intra cluster node. So, total area covered by cluster head node and how far the base station locates from the CH doesn't take into account. Table-1, shows the parameter which we take into account for our simulation.

Table 1. Simulation Parameter

Parameter	*Value*
Node's energy	500-600Jules
Number of nodes	50
Round time	50 Seconds

Now, at first see the mathematical calculation. Let, taking 50 sensor nodes which mention in table.

In traditional case, total wake up time of CH = Total Round time.

In Proposed case, total wake up time of CH $=\sum_{i=0}^{n}(Preamble\ Time + Data\ Processing\ Time)$

In Normal case its consumed energy = 50 Jules [Taking into consideration, per node consumed 1 Joule per round]

In Proposed case, if 25 nodes out of 50 nodes wants to transmit data to the CH,
Then, total consumed energy = 25 Jules [per node consumed 1 Joule per round] + 6.25 Jules [preamble consumed energy=.25 Jules]

= 31.25 Jules.

Here it saves 18.78 Jules in this current round. It saves much energy in low traffic.

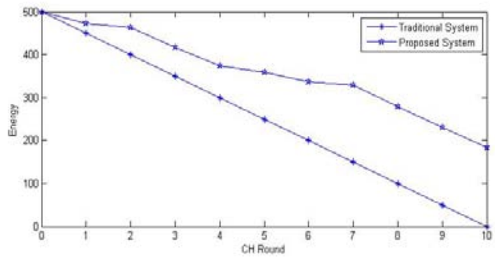

Figure 5. CH Round Vs Energy

Figure-5 shows the discharging non cluster head node. Here, X-axis shows CH round and Y-axis shows Energy level. Figure shows that, normal case discharge its energy after 10 rounds [Only transmission energy taking into account, other factors are negligible, we avoid it. That's why curve look like linear]
Whereas, in our proposed system node contain more than 180 Jules after ten rounds.
On the other hand, Figure- 6 shows the comparative analysis of transmission delay of intra cluster nodes. X-axis contains the number of intra cluster node and Y-axis contains the Latency.

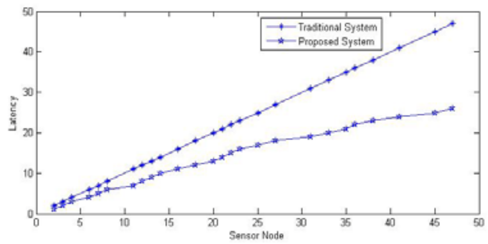

Figure 6. Sensor Node Vs Latency

VI. CONCLUSION AND FUTURE WORK

This paper proposed the enhancement of existing LEACH protocol. According to our simulation result, our proposed method gives better result in low traffic. It will decrease latency of sensor node to send data to the cluster head. Node need not wait for its time period to send data to cluster head.

Future work includes system scaling studies and parameter analysis. More tests will be done on larger test beds with different number of nodes.

REFERENCES

[1] W. Ye, J. Heidemann, and D. Estrin. An energy-efficient MAC protocol for wireless sensor networks. In 21st Conference of the IEEE Computer and Communications Societies (INFOCOM), volume 3, pages 1567{1576, June 2002. **(Conference)**

[2] Ilker Demirkol, Cem Ersoy, and Fatih Alagöz, Bogazici University, "MAC protocols for wireless sensor networks: a survey" IEEE Communications Magazine, vol. 44, no. 4, pp. 115-121, Apr. 2006. **(Journal)**

[3] LANMAN Standards Committee of the IEEE Computer Society, Wireless LAN medium access control (MAC) and physical layer (PHY) specification, IEEE, New York, NY, USA, IEEE Std 802.11-1997 edition, 1997.

[4] Holger Karl, Andreas Willig, "Protocol and Architectures for Wireless Sensor Networks", John Wiley & Sons, 2005, pp.110-148. **(Book)**

[5] Tijsvan Dam, Koen Langendoen, "An Adaptive Energy-Efficient MAC Protocol for Wireless Sensor Networks", SenSys '03, pp. 171-180. **(Conference)**

[6] Y.Z.Zhao, C.Y.Miao, M.Ma, "An Energy-Efficient Self-Adaptive Duty Cycle MAC Protocol for Traffic-Dynamic Wireless Sensor Networks" February 2013, Volume 68, Issue 4, pp 1287-1315 **(Journal)**

[7] Suresh Singh, C.S. Raghavendra, "PAMAS - Power Aware Multi-Access protocol with Signalling for Ad Hoc Networks", Pages 5 -26 . **(Conference)**

[8] Yanjun Sun, Shu Du, Omer Gurewitz, and David B. Johnson. "DW-MAC: A low latency, energy efficient demand -wakeup mac protocol for wireless sensor networks".pp. 53-62 MobiHoc 2008. **(Conference)**

[9] Y.Z.Zhao a, M.Ma b, C.Y.Miao a, T.N.Nguyen, "An energy-efficient and low-latency MAC protocol with Adaptive Scheduling for multi-hop wireless sensor networks", Volume 33, Issue 12, 15 July 2010, Pages 1452–1461 **(Journal)**

[10] V. Rajendran, K. Obraczka, J.J. Garcia-Luna-Aceves, "Energy-Efficient, Collision-Free Medium Access Control for Wireless Sensor Networks", Proc. ACM SenSys 03, Pages:181 - 192, Los Angeles, California, 5-7 November 2003. **(Conference)**

[11] M. J. Handy, M. Haase, D. Timmermann, "Low Energy Adaptive Clustering Hierarchy with Deterministic Cluster-Head Selection" Fourth IEEE Conference on Mobile and Wireless Communications Networks. **(Conference)**

[12] A.El-Hoiydi, J,-D.Decotignie, "Wise MAC: An Ultra-Low Power MAC Protocol for the Down link of Infrastructure Wireless Sensor Networks", Computers and Communications, 2004 ,ISCC, vol-1, pp. 244 – 251 **(Conference)**

[13] Joseph Polastre, Jason Hill, David Culler, "Versatile Low Power Media Access for Wireless Sensor Networks", SenSys '04 pp. 95-107 **(Conference)**

Software Defined Radio for PC to PC Data Communication

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***Abstract*—A combination of hardware and software technologies where some or all of the radios operating functions are implemented through adjustable software or firmware operating on programmable processing technologies is defined as Software Defined Radio (SDR). The basic concept of SDR is that the radio functions are configured by software and number of areas can be covered by using same platform. As software is used, configurations can be changed according to various radio functions which are not captured by classic radio. The tasks to be performed included the channels configuration, the management of the data transfer between two PC, the baseband data modulation and demodulation, and the data organization into packets solely by software. For the physical layer of this system Orthogonal Frequency-Division Multiplexing (OFDM) is chosen as the transmission multiplexing method. This choice has been made because of the advantages that OFDM has better channel capacity and provides larger data rates. In this paper, simple IR circuit is used as SDR platform. This IR circuit work as a transceiver which can transmit text, number or image from one PC to another PC. In order to verify the proper functionality of the communication scheme, the received data streams are further analyzed with the use of MATLAB.**

Keywords: Software Defined Radio; Orthogonal Frequency Division Multiplexing; IR Interfacing; Data Communication.

I. INTRODUCTION

A Software Defined Radio is a system that transmits or receives signals in the radio frequency (RF) part of the electromagnetic spectrum for the purpose of transferring information using same platform. The SDR approach is helpful because there is a scope of developing a system which is compatible with more than one mobile communication standard. This can be achieved by using reconfigurable software for different technologies. Software Defined Radios (SDRs) are driving the integration of digital signal processing (DSP) and radio frequency (RF) capabilities. This integration allows software to dynamically control communications parameters such as the frequency band used, filtering, modulation type, data rates and frequency hopping schemes.

Traditional hardware based radio devices can only be modified through physical intervention, which results in higher production costs. Moreover in case of supporting multiple waveform standards it is less flexible. On the other hand SDR technology is more efficient and cost effective, which allow multiform and multi-functional wireless devices that can be enriched by modifying software program [1].

II. ORTHOGONAL FREQUENCY DIVISION MULTIPLEXING (OFDM)

In OFDM technique, input bit stream is divided into several parallel bit streams which are used to modulate several subcarriers. A guard band separates these sub-carriers to avoid overlapping with each other. In OFDM receiver side, no bandpass filter is required to separate the spectrum of individual sub carriers, because of the orthogonality nature of the subcarriers. This orthogonality is achieved by performing Fast Fourier Transform (FFT) on the input bit stream [2]. “Fig.1” shows the comparison between conventional FDM and OFDM.

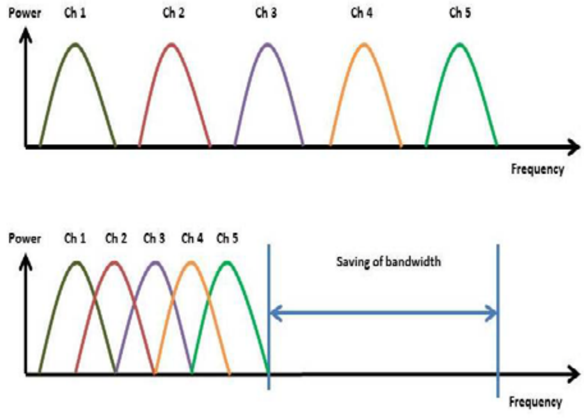

Figure 1: Comparison between conventional FDM and OFDM.

A. OFDM Generation and Reception

“Fig.2” shows the basic block diagram of OFDM transmitter and receiver. At transmitter side, frequency domain based source symbols are required for OFDM system. These symbols are feed to IFFT block to convert them into time domain. Suppose N numbers of sub-carrier are chosen then the function of IFFT are N orthogonal sinusoids of different frequency, finally receive N symbols at a time. Each of N complex valued input symbols determines both the amplitude and phase of the sinusoid for that subcarrier. These all N sinusoids make up a single OFDM symbol that is output of IFFT block [3].

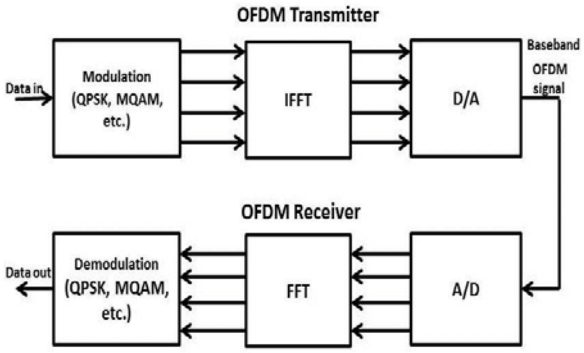

Figure 2: Basic OFDM transmitter and receiver.

At receiver side, FFT block is used to perform the reverse operation which makes the received signal back into frequency domain[4].

B. *Advantages of OFDM technique over other techniques*

Though at present many modulation and demodulation technique are available, we choose OFDM technique because of some additional benefits over other techniques. OFDM technique shows high spectral efficiency because of overlapping spectra which is found by orthogonality nature. It provides low receiver complexity as the transmitter combat the channel effect to some extends. It has a low complexity multiple access schemes such as orthogonal frequency division multiple access (OFDMA). Moreover, it is suitable for high data rate transmission [5].

C. *Simulation outcome*

"Fig.3" shows the transmitted OFDM signal.

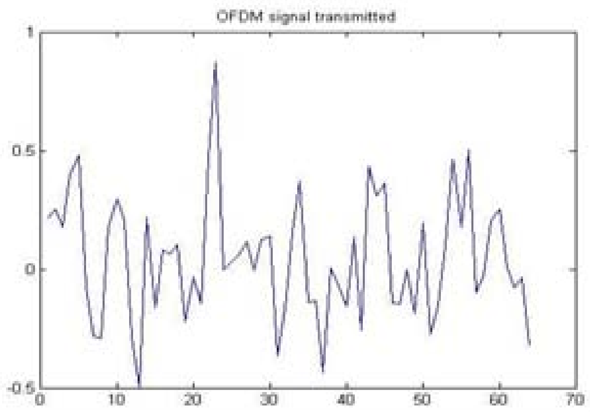

Figure 3: Transmitted OFDM signal.

"Fig.4" shows the simulation outcome of OFDM signal (power spectral density vs. frequency).

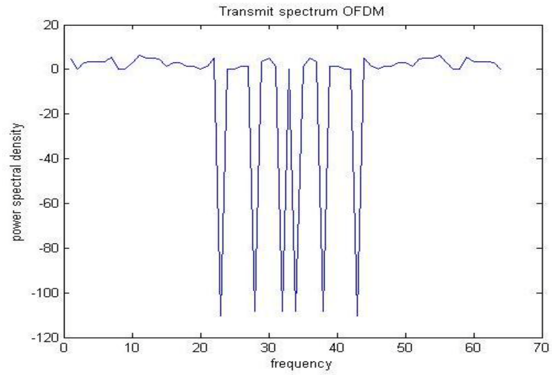

Figure 4: Transmitted OFDM spectrum.

"Fig. 5" shows the received OFDM signal without noise.

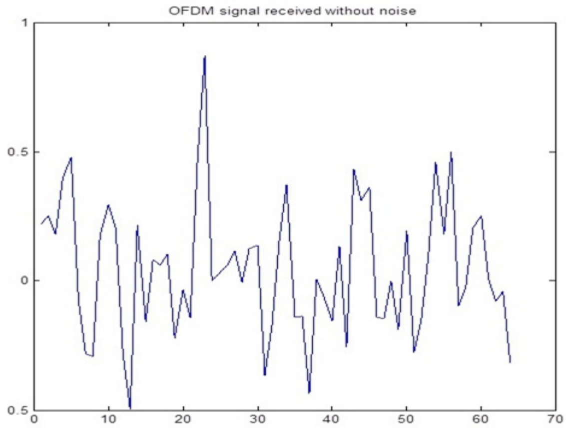

Figure 5: Received OFDM signal without noise.

"Fig. 6" shows the received OFDM signal with AWGN noise.

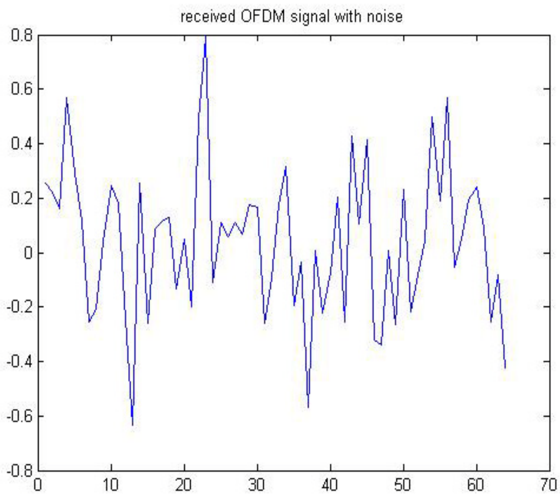

Figure 6: Received OFDM signal with noise.

"Fig. 7" shows BER vs SNR curve.

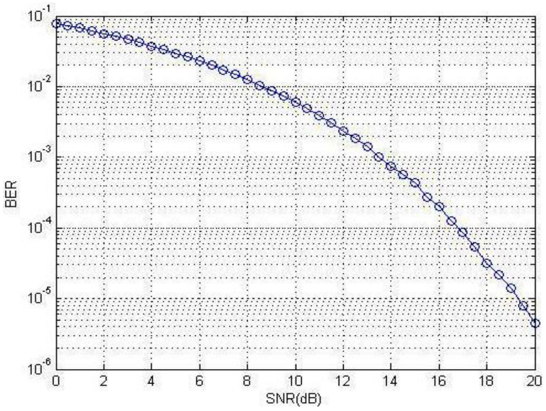

Figure 7: BER vs SNR curve.

III. DESIGN AND FABRICATION

Various methods of data communication are available all over the world. Some of the method show higher data rates, some show better channel capacity, but in this paper the combination of all expected data transfer characteristics are highlighted by using simple IR circuit. Two IR circuits are used; one for sending and one for receiving. Each of the circuit contains:

- Voltage regulator (LM-7805)
- Resistor (1k ohms)
- Capacitor (10mikroferad)
- Crystal oscillator (8MHz)
- Micro-controller (PIC-18F2550)
- MAX-232
- Breadboard
- IR emitter
- IR receiver
- LED
- Power supply
- USB converter (U-232)
- Connecting wire

A. *Implementation*

At first output of the voltage regulator (LM-7805) is connected to the positive pin of power supply. Then input voltage regulator (LM-7805) is connected to power line. A resistance of 1kΩ is connected to the power line with a LED to check whether the circuit gets power. The pin number of 13 of PIC-18F2550 is connected to IR emitter which is connected to a resistance of 1kΩ; this 1k ohm resistance is further connected to ground. IR emitter is used for data transmit [6, 7].

The pin number of 3 of PIC-18F2550 is connected to one of the pin of IR receiver. Second pin of IR receiver is connected to power line. Third pin is connected to ground. IR receiver is used for data receive.

The pin number of 1 of PIC is connected to a resistance of 1k ohm which connected to power line. The pin number of 9 and 10 of PIC is connected to 8MHz crystal oscillator. This is used to give oscillation. Voltage is supplied to 23 pin and 24 pin is grounded of PIC.

The output pin of 25 and 26 of PIC are connected to input pin of 9 and 10 respectively of MAX-232. Voltage is supplied at 16 pin and 15 pin is grounded of MAX-232. There are four capacitors (10□F) is connected to MAX-232 which is shown in our diagram. The pin number of 7 and 8 are to USB converter (U-232) [8].

"Fig.8" shows IR transceiver for PC1 that works as either transmitter or receiver. As two PC is needed for data communication. Both PC must have the IR circuit; one for data sending purpose and another for data receiving purpose and vice-versa.

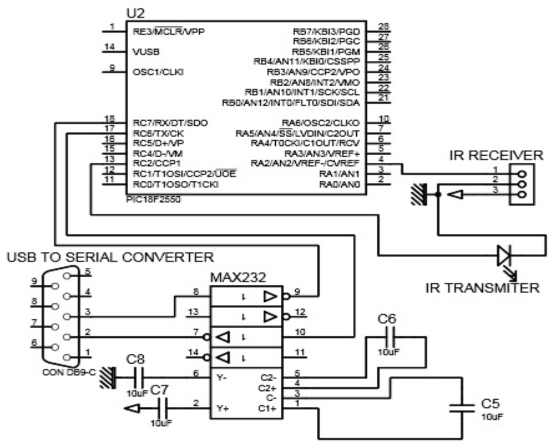

Figure 8: Schematic diagram of transceiver for PC1.

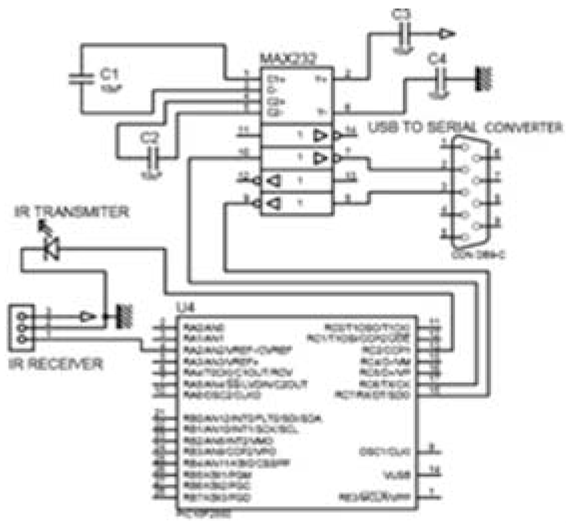

Figure 9: Schematic diagram of transceiver for PC2.

"Fig.9" shows another IR transceiver that works as either transmitter or receiver.

B. *Working Principle*

At first voltage is supplied to the network diagram.USB converter (U-232) is connected to PC. Data is sent through IR emitter which is received by IR detector. In this way data communication is performed from one PC to another PC. "Fig.10" shows the snap of simple IR circuit as a SDR platform.

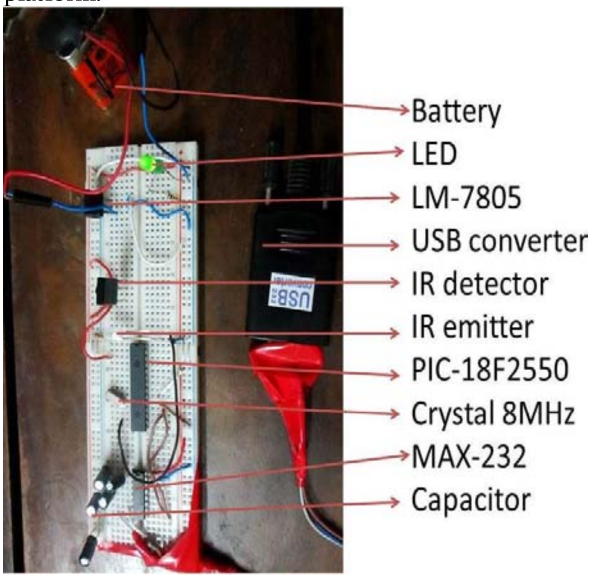

Figure 10: Snap of the IR circuit.

Designing a proper Data Communication System network and implement its function will give a good output of any system and using it for many purposes in networking activity.

IV. METHODOLOGY OF DATA COMMUNICATION

A. *Steps of Data Transmit*

- Connect the USB to USB port of one PC.
- Check the serial port number of that pc from"mycomputer/device manager/port".
- Run mikroC.
- Go to tool bar;"tools/USART terminal".
- Select COM port from"COM port setting" tab.
- Choose any"data format".
- Then press"connect" from"command" tab.
- Type"S" two times; be sure that whether the systemis in sending mode.

Desired data transmission can be performed by running MATLAB program.

B. *Steps of Data Reception*

Data reception procedure is similar to data transmission procedure except the last step. In this case:

- Type 'R' two times for checking that whether the system is in receiving mode.
- Finally receive the expected data on mikroC.

Desired data can also be received by running the"receiving"program on MATLAB.

V. BENEFITS

A. *Benefits of SDR over Conventional Radio*

As more and more of the conventional hardware circuitry is being replaced by software, SDR receiver's functionality can be changed by software upgrade only. Thus affords far greater flexibility and reliability to the designer. Parameters of hardware components are subject to temperature changes, manufacturing variations and aging. However, software always performs the same. This is why SDR exhibits far better parameter predictability and performance consistency. Hardware products are hard and expensive to improve and upgrade. In contrast, SDR products are to a significant degree future proof and can be improved by a simple software upgrade only, with minimum equipment downtime. Due to a reduced number of hardware parts and software reusability, an SDR product is easier and cheaper to manufacture and maintain. Moreover, SDR also shows reusability, reconfigurability and enhanced functionality features which are not found in conventional radio.

In word, SDR can be flexible enough to avoid the "limited spectrum" assumptions of designers and it provides more easy inter-operability [9].

B. *Benefits over Other System of Data Communication*

Data transfer by using simple IR circuit has additional benefits in this respect.

- Spectrum analysis
- Higher data rates
- Real time data transfer
- Better channel capacity
- Wide range networking
- Reliable and modifiable
- Low cost
- Easy inter-operability

Moreover, if IR can be replaced by antenna a certain area is being under the network coverage.

VI. Experimental Result

Experimental result is shown in “Table 1” where different bits are transmitted and received. Comparisons between them are also shown in this Table.

Table 1: Comparative Results of Different Data Communication

Parameters / No. of bits	Time Taken According to Distances(sec.)			Data Rates (app.)
	0.5m	1m	2m	
100	6.5	6.5	7	15.02bps
400	27	27	27.5	14.72bps
500	34.5	34.5	35	14.42bps
1000	69.5	69.5	70	14.36bps
2000	140.5	140.5	141	14.21bps

“Table 2” shows the variation of bit error in percentage according to distances in meter for 5000 bits.

Table 2: Variation of Bit Error according to Distances for 5000 bits

Distance (meter)	0.25	0.50	0.75	1.00	1.25	1.50	1.75	2.00
Percentage of Error (%)	0.02%	0.02%	0.04%	0.1%	0.12%	0.14%	0.18%	0.18%

“Fig. 11” shows graphical representation of table II. This curve is a non-linear curve. From this curve we come to a decision that percentage of error is non-linearly increase with the increase of distances.

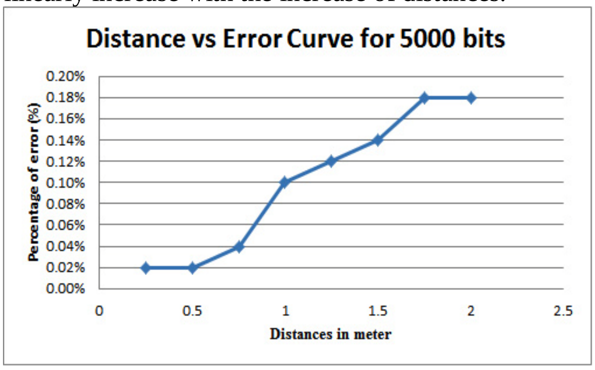

Figure 11: Percentage of Error vs Distance Curve for 5000 bits

“Table 3” shows the variation of bit error in percentage according to deviations in degree for 5000 bits.

Table 3: Variation of Bit Error according to Deviations for 5000 bits

Deviation of line of sight (degree)	0	2	4	6	8
Percentage of Error (%)	0.02%	0.08%	0.18%	0.26%	0.4%

“Fig.12” shows graphical representation of “Table 3”. The curve is a non-linear curve. From this curve we come to a decision that percentage of error is non-linearly increase with the increase of deviations of line of sight. And gradually no data is transmitted and received if the deviation is far greater than 8 degree or non line of sight.

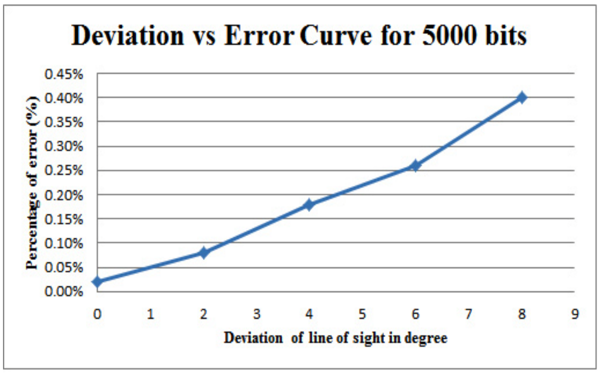

Figure 12: Percentage of Error vs Deviation Curve for 5000 bits

VII. Future Trends

The Data communication system using simple IR circuit can be further modified and implemented. Some upcoming improvements are as follows.

- Smart networking system
- RFID Application
- Management of Information Flow
- Smart Calibration
- Smart Bridge (recognize the need for two or more legacy to communicate and connect them through a bridge)
- Data rates for correct processing of radio signals
- Development of software technologies, platforms and tools
- Applications of Spread Spectrum in SDR systems [10].

VIII. Conclusion

In this paper, we proposed and implemented a reconfigurable SDR platform by combining it with data communication system using simple IR circuit. Furthermore, realization of digital data

communication is achieved by applying SDR approach. SDR approach avoid the radio interference, which causes lower transmit performance, and provide an efficient wireless digital communication.
In word, SDR is a promising technology that facilitates development of multi-band, multi-service, multi-standard, multifeature data communication and future-proof network infrastructure.

This paper shows the feasibility of implementation and the performance of a Software Defined Radio data link, using two PC and a high level programming language.

REFERENCES

[1] (2013, Aug.) The sdrforum website. [Online]. Available: http://www.sdrforum.org/pages/document

[2] Hasan, M.A. "Performance Evaluation of WiMAX/IEEE 802.16 OFDM Physical Layer," Helsinki University of Technology.

[3] Hasan, M.A. "Performance Evaluation of WiMAX/IEEE 802.16 OFDM Physical Layer," Helsinki University of Technology.

[4] Hasan, M.A. "Performance Evaluation of WiMAX/IEEE 802.16 OFDM Physical Layer," Helsinki University of Technology.

[5] Hasan, M.A. "Performance Evaluation of WiMAX/IEEE 802.16 OFDM Physical Layer," Helsinki University of Technology.

[6] (2013, Sep.) Data sheet of LM-7805. [Online]. Available: http:// www.Data sheet reference.com

[7] (2013, Sep.) Data sheet of PIC-18F2550. [Online]. Available: http:// www.Data sheet reference.com

[8] (2013, Sep.) The wikipidea website. [Online]. Available: http://www.wikipedia.org/wiki/MAX-232

[9] (2013, Sep.) The wikipidea website. [Online]. Available: http://www.wikipedia.org/wiki/Softwaredefined radio

[10] Chen, C.Y., Tseng, F.H., Chang, K.D., Chao, H.C. and Chen, J.L., "Reconfigurable Software Defined Radio and Its Application," Taiwan

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Linear Polarization Switchable Patch Array Antenna using Magic-T Bias Circuit and Orthogonal Feed

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Abstract— **In this paper, a linear polarization switchable patch array antenna is proposed. The orthogonal feed circuit and magic-T circuit is introduced to realize the proposed array antenna. The advantage of the magic-T circuit is the excellent isolation between the RF signal and the switching bias signal. The microwave integration technology is effectively employed to realize proposed linear polarization switchable array antenna. The proposed array antenna consists of four patch elements and 16 PIN diodes. In order to realize the ±45º polarization switching, four switching diodes are integrated with each patch elements. Using the ON/OFF condition of the diodes, the polarization axis can be easily switched to ±45º. The array antenna is realized in very simple and compact structure as all the antenna elements, feeding circuit and bias circuit are arranged on both sides of a dielectric substrate. The ability of the proposed array antenna to switch the polarization axis at ±45º at 10 GHz (X band) is confirmed by the experimental investigation.**

Keywords*— *Both-Sided MIC technology, Orthogonal polarized array antenna, Polarization Switching, Magic-T.

I. INTRODUCTION

Along with the rapid development of the wireless communication systems, the planar antenna technology is emerging and being widely used in various sectors of wireless communications systems due to their low profile, light weight, low cost and easy integration with active components [1, 2]. In the past decades, the planar antenna technologies had an enormous development in reconfigurable, compact, broadband, gain enhancement and so on to meet the requirements of the wireless communications and the ubiquitous society. Moreover, the evolution of the planar antenna technology keeps emerging. In addition, the integrated and active integrated antennas receive a great deal of attention because they can reduce the size, weight and cost of many transmit and receive systems [3]. Loading of circuit components such as semiconductor devices and various kinds of IC's in microwave resonators or antennas to build up the purposeful electromagnetic field on them, is referred as the Microwave Integration Technology [4]. There are some reports on this concept of Microwave Integration Technology [5-7]. There is a report on a polarization switchable slot antennas [8]. The authors of paper [8] have reported a 4-element slot-ring array antenna where 4 PIN diodes are loaded on each slot ring antenna totaling 16 diodes. However, the bias signal is superimposed with the RF signal to the co-axial feed. As a result, the cross-polarization isolation of the radiation pattern of the array antenna was not achieved better than -15 dB. For the proposed array antenna, the bias circuit is arranged with the feed line by using the magic-T junction. In this case the cross-polarization isolation is significantly improved to better than -20 dB. Momentum of the Advanced Design System (ADS, Agilent Technologies) is used for the simulation of the feed circuit and the bias circuit and the array antenna elements. The ON state diodes are replaced by short circuit for the experimental investigation.

II. THE PROPOSED ARRAY ANTENNA

Fig. 1 shows the schematic structure of the proposed array antenna. The Both-sided MIC technology [9-11] is effectively employed to realize the array antenna. The array antenna consists of four patch elements and 4 switching diodes are loaded on each patch elements which totals in 16 diodes. The feed and bias circuits are arranged by using the magic-T and air bridge as seen in the dotted square of Fig. 1. The antenna elements, diodes microstrip lines are arranged on the obverse side and the slot line is arranged on the reverse side of Teflon glass fiber substrate. The design of the magic-T and the antenna elements will be explained in this section.

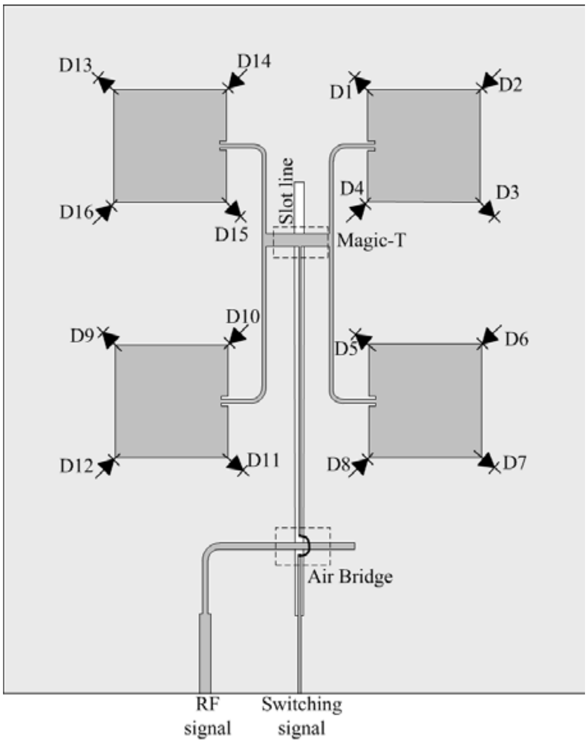

Figure 1. The proposed array antenna

A. *The Magic T*

Planar magic-Ts are used in microwave integrated circuits to split or combine in-phase or anti-phase signals. The magic-T can be very useful for balanced-mixers, discriminators, and beam-forming networks. The advantages of the magic-T are low insertion loss, high isolation, compact size, and fabrication simplicity [12]. There are some reports for realizing the magic-T using the co-planar waveguide (CPW) or microstrip (MS) to slot line (SL) mode conversion techniques [13-16].

Fig. 2(a) shows the structure of the magic T. Fig. 2(b) represents the RF and bias signal modes for the microstrip and slot lines. The RF signal propagates by the odd mode and the bias signal propagates by the even mode. When the RF signal is input to the port 1, the RF signal is split to the microstrip line in anti-phase and propagates to port 2 and port 3, which is known as the odd mode. The bias signal from port 4 can propagate to port 2 and port 3 by the means of even mode. As the even mode and the odd mode are orthogonal to each other, a good isolation between the RF signal and the bias signal can be realized. Fig. 3 shows the simulated S-parameter for the magic-T structure. The S_{11} for the structure is matched at the design frequency of 10 GHz. S_{41}, which is the isolation, also achieved better than -28 dB. In addition, the S_{21} and S_{31} are achieved around 4 dB for the simulation.

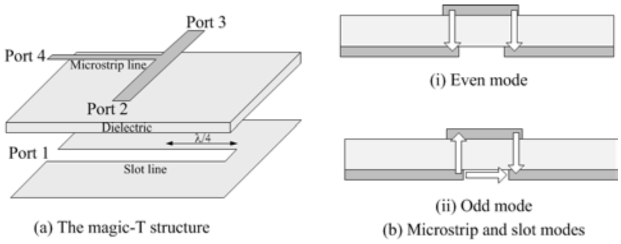

Figure 2. The magic-T structure

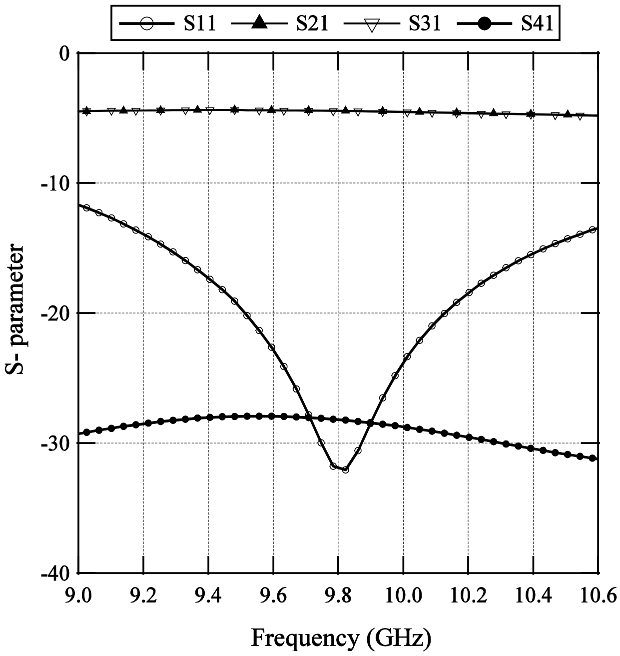

Figure 3. Simulated S-parameters of the magic-T structure

B. *Basic Behavior of the array antenna*

The basic behavior of the array antenna is explained in this section. The RF signal is fed to the antenna elements in same phase because of the parallel feed circuits. The switching bias signal is fed to the antenna elements through the microstrip line connected with the feed line in magic-T. Depending on the polarity of the bias voltage, the diodes of the antenna elements becomes ON/OFF.

When positive bias voltage is applied to the bias line, the odd numbered diodes D1, D3, D5, D7, D9, D11, D13 and D14 becomes ON due to the forward bias condition and the even numbered diodes D2, D4, D6, D8, D10, D12, D14 and D16 remains OFF state due to the reverse bias condition. In this case, the surface current of the array antenna elements flows as shown in Fig. 4(a). And in this case, the polarization axis is tilted to -45º. Same behavior can be applicable for the negative bias condition as shown in Fig. 4(b). And in this case, the polarization axis is tilted to +45º.

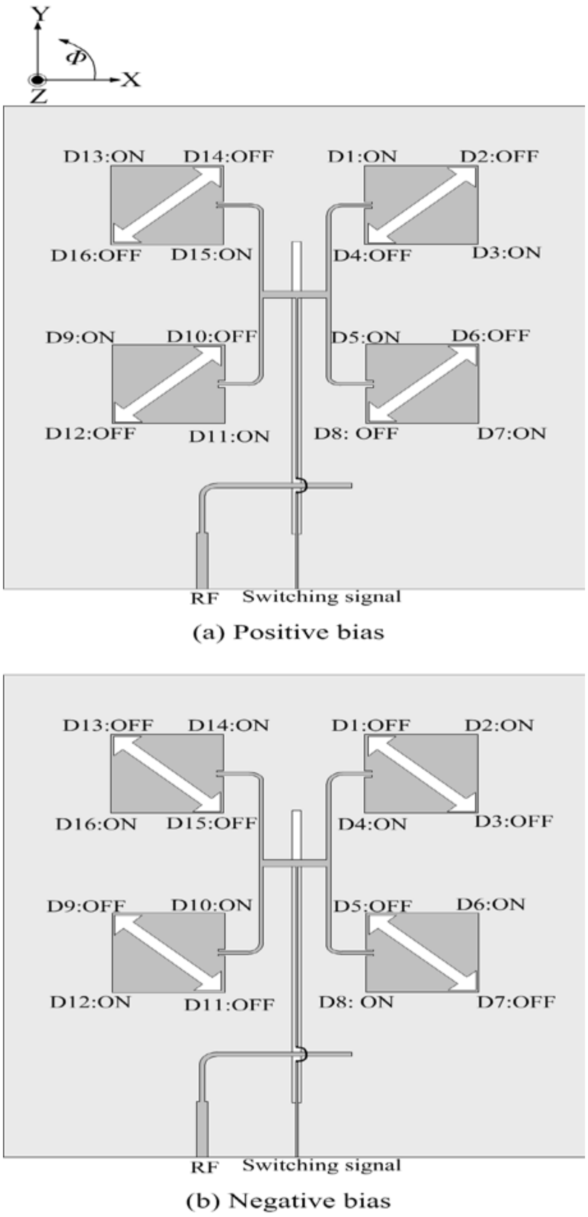

Figure 4. Basic behavior of the array antenna

III. DESIGN OF THE ARRAY ANTENNA

Fig. 5 shows the dimension of the proposed array antenna. The size of the antenna element is 9.23 mm×9.23 mm at the design frequency of 10 GHz. Four PIN diodes are loaded on four corners of each antenna element in order to realize polarization switching. The other ends of the diodes are connected with the ground plane by conductor through via hole. The characteristic impedance of the microstrip lines connected with the antenna elements is 110 Ω and the width of the microstrip line is 0.55 mm. The input impedance of antenna element is adjusted to the microstrip lines by properly inserting a pair of notches at the patch [17-18]. The width of the notch is 0.2 mm and length is 0.4 mm. The 110 Ω microstrip lines connect with another microstrip line which is used to realize the Magic-T. Impedance of this microstrip line is 55 Ω and the width is 2.4 mm. The width of the bias line is maintained at 0.2 mm in order to achieve a high impedance of 154 Ω. A slot line of 0.2 mm width and length of 30 mm is arranged exactly below the bias line on the ground plane of the array antenna. The bias line is connected with the bias port through air-bridge. A 110 Ω RF microstrip line is arranged on the obverse plane upon the slot line as a microstrip-slot parallel branch circuit. This RF microstrip line is connected with the 50 Ω RF port using a quarter wavelength ($\lambda_g/4$= 5.55 mm) impedance transformer whose impedance is 71 Ω and width is 1.36 mm. The size of the Teflon glass fiber substrate and the ground plane is 48 mm × 54 mm. The thickness of the substrate is 0.8 mm with the relative dielectric constant of 2.15.

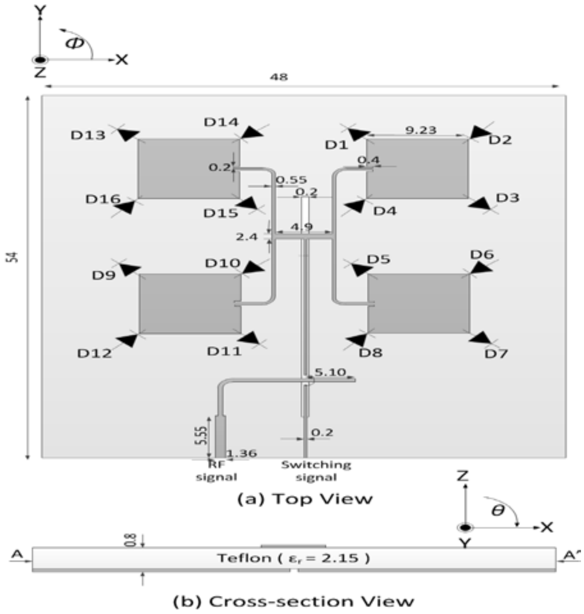

Figure 5. Dimension of the proposed array antenna (scale in mm)

IV. EXPERIMENTAL RESULTS AND DISCUSSION

After you As already mentioned the ON state diodes are replaced by the short circuit and the OFF state diodes are replaced by the open circuit for the experiment of the array antenna. The experiment was performed in an anechoic chamber. Impedance matching is the most important issue for the design of the array antenna which is represented by the s-parameters. Fig. 6 shows the experimental result for the S-parameter characteristic of the array antenna. For both positive and negative bias voltage, the S_{11} better than -20 dB is achieved at 10 GHz which proves the feed circuit is matched with the array antenna at the design frequency. The isolation of the switching signal and the RF signal was also achieved better than -40 dB, which is a very good result. Fig. 7 shows the experimental results of the frequency characteristics of the polarization angle of the array antenna. The polarization angle is around +45º for the antenna. The radiation pattern is the measure of the signals radiated by the antenna. The radiation pattern measured at the design frequency of 10 GHz is shown in Fig. 8. The cross polarization isolation of better that -20 dB is achieved. Therefore, compared with [8], the proposed array antenna confirms a better performance of the array antenna with better cross-polarization isolation.

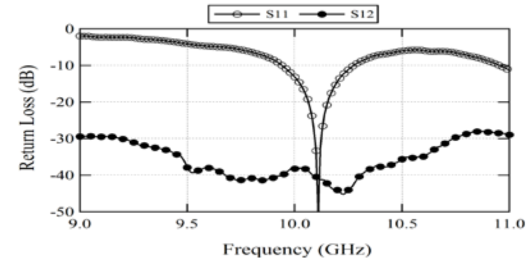

Figure 6. Experimental S-parameter for the array antenna

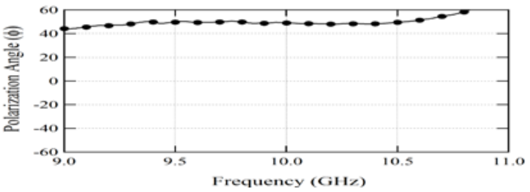

Figure 7. Experimental result for the polarization angle

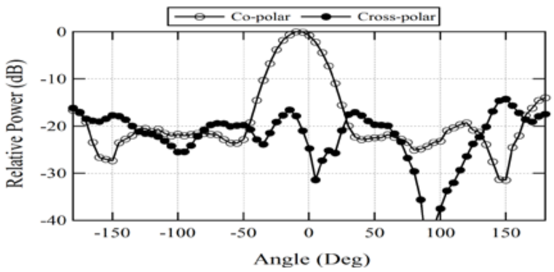

Figure 8. Experimental result for the radiation pattern of the array antenna

V. CONCLUSION

In this paper, a linear polarization switchable patch array antenna is proposed. The orthogonal feed circuit and magic-T bias circuit is used to realize the array antenna. The microwave integration technology is effectively employed to realize proposed linear polarization switchable array antenna. In order to realize the ±45º polarization switching, two switching diodes are integrated with each patch element. Using the ON/OFF condition of the diodes, the polarization axis can be easily switched to ±45º. The experimental results confirm the ability of the proposed array antenna to switch the polarization axis at ±45º at 10 GHz (X band). The proposed array antenna confirms some better performances compared with [8] which are listed below:

I) Antenna configuration become more simple

II) The cross-polarization isolation of radiation pattern is significantly improved to -20 dB which was -15 dB for [8].

The proposed array can be an attractive candidate for the application of polarimetric sensors, polarization diversity and wireless data transmission etc.

REFERENCES

[1] E. Nishiyama, M. Aikawa, and S. Egashira, “FDTD analysis of stacked microstrip antenna with high gain”, Progress in electromagnetic research, PIER 33, 29-43, 2001.

[2] Yang F. and Rahmat-Sami, “Patch Antennas with Switchable Slots (PASS) in Wireless Communications: Concepts, Designs, and Applications”. IEEE Antennas and Prop. Vol.47, No.2, April 2005. PP.13-29.

[3] Qian Y., Itoh T., “Progress in active integrated antennas and their applications”, IEEE Trans. Micro. Theory Tech., 1998, 46, (11), pp. 1891-1900.

[4] M. Aikawa, E. Nishiyama, T. Tanaka., “Advanced Utilization of resonant fields and its applications to push-push oscillators and Reconfigurable Antennas”. IEICE Trans. Electron. Vol. E89-C, No. 12, PP. 1798-1805, December, 2006.

[5] M. Aikawa, ‘Technical approach for advanced microwave circuits and antennas-an approach on microwave integration’, IEICE Technical report, MW2003-202, Nov. 2003.

[6] M. Aikawa, ‘Prospect of microwave engineering: Evolution for adaptive microwave circuits’ , J. IEICE, Vol. 83, No.8, pp. 595-599, Aug 2000.

[7] M. Aikawa, E. Nishiyama, T. Tanaka, ‘Evaluation integration approach for microwave planar circuits’, IEICE Trans. Electron (Japanese Edition), Vol. J89-C, No.5, pp. 183-190, May 2006.

[8] E. Nishiyama M. Aikawa S. Sasaki, “Polarisation switchable slot-ring array antenna” IET Microw. Antennas and Propag., 2008, Vol. 2, No. 3, pp. 236–241

[9] M. Aikawa, H. Ogawa, “Double-Sided MIC’s and Their Applications,” IEEE Trans. Microwave Theory Tech., vol.37, no.2, pp.406-413, Feb. 1989.

[10] K. Egashira, E. Nishiyama, and M. Aikawa, “Microstrip Array Antenna using both-sided MIC feed circuits”, Asia-pacific Microwave conference, proc, 2002.

[11] M. Aikawa and H. Ogawa, “A new MIC magic-T using coupled slot lines,” *IEEE Trans. Microw. Theory Tech.*, vol. MTT-28, no. 12, pp. 523–528, Dec. 1980.

[12] Kongpop U-yen, Wollack, E.J., Moseley, S.H., Papapolymerou, J., Laskar, J., “A Compact Low-loss Magic-T using Microstrip-Slotline Transitions”, IEEE/MTT-S International Microwave Symposium, pp. 37 – 40, 2007

[13] C.-H. Ho, L. Fan, and K. Chang, "New uniplanar coplanar waveguide hybrid-ring couplers and magic-T's," IEEE Trans. Microwave Theory & Tech., vol. 42, no. 12, pp. 2440-2448, December 1994.

[14] J. P. Kim and W. S. Park, "Novel configurations of planar multilayer magic-T using microstrip-slotline transitions," IEEE Trans. Microwave Theory & Tech., vol. 50, no. 7, pp. 1683- 1688, July 2002.

[15] M. Aikawa and H. Ogawa, "A new MIC magic-T using coupled slot lines," IEEE Trans. Microwave Theory & Tech., vol. 28, no. 6, pp. 523-528, June 1980.

[16] T. Hiraoka, T. Tokumitsu and M. Aikawa, "Very small wideband MMIC magic-T' s using microstrip lines on a thin dielectric film," IEEE Trans. Microwave Theory & Tech., vol. 37, no. 10, pp. 1569-1575, October 1989

[17] R. E. Munson, “Conformed microstrip antennas microstrip patched arrays,” IEEE Trans. Antennas Propagat., vol. AP-22, pp. 74–78, 1974.

[18] P. S. Hall and C. M. Hall, “Coplanar corporate feed effects in microstrip patch array design,” Proc. Inst. Elect. Eng. , pt. H, vol. 135, pp. 180–186, 1988.