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on
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And Management
(March 28-29’ 2014)

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Tribute to
Late Shri G.V.K.Sinha
Founder Chairman, Lingaya's Group

"The best way to predict the future is to create...
When a collection of brilliant minds,
Hearts and talents come together...
Expect a masterpiece"

Prof. G.V.K. Sinha, a unique educationist with remarkable insight into technical education and great expertise in managing human resources was a self-made icon in the field of technical and management education. With the B.Tech Degree he entered his career as a lecturer, climbed up the ladder as head of the Department, Asstt. Director and then the Principal in various reputed technical institutions in New Delhi, and later full-time consultant of AICTE, New Delhi. Everywhere Prof. Sinha dedicated himself to explore and contribute the best for the noble cause of education. With grace and humanitarian values finally Prof. Sinha reached to the peak of his career as the Chairman of Lingaya’s Group, set-up by him in memory of his father, the great freedom fighter Gadde Lingaya. His expertise in organizing skill in curriculum review and efficiency in training and placement enabled him to make LIMAT the most luring educational institution for the aspiring professionals in Delhi NCR. LIMAT is the living legend of Prof. Sinha’s noble goal and higher vision in the field of Education. In recognition to his excellent achievements in the field of Technical education, the LIMAT was upgraded to Lingaya’s University with approvals from MHRD, Govt. Of India, w/s 3 of UGC Act 1956.

Prof. Sinha left us for the heavenly abode on 6th June, 2011. May the departed soul rest in peace?

"Par Excellence with Human Touch"
Dr. Picheswar Gadde  
CEO,  
Lingaya's University

Message

It gives me immense joy to know that the school of Computer Sciences and Engineering is organizing the International conference on Data Acquisition, Transfer, Processing and Management on March 28-29' 2014

Because of the recent developments in the areas of Information technology, it has percolated in our day to day life to the extent that we get up in the morning, checking our mail and before going to bed check our mails again. During the day time, we keep in touch with the mail and various other forms of information technology, without even knowing that we are using the Information Technology in sorting out the day to day issues.

During the usage of the Information Technology, correctness and promptness is required in each phase of data acquisition, transfer, processing and management. I sincerely hope that the event will provide an appropriate platform for one and all.

I welcome the keynote speakers, faculty members and research scholars from around the world to the conference and sincerely hope that the conference will set a benchmark for research in the fields of Data Acquisition, Transfer, Processing and Management.

I also congratulate school of computer sciences and Engineering of Lingaya's University, Faridabad and organising committee of the conference for putting together their efforts in pursuit of excellence in research and academics
Message

I feel delighted in extending a warm welcome to all the keynote speakers & delegates of International Conference on Data Acquisition, Transfer, Processing & Management organized by School of Computer Sciences, Lingaya’s University, Faridabad on March 28-29, 2014.

The theme of the Conference selected by School of Computer Science is very appropriate and suites the contemporary work in the area of IT and its applications. This Conference is quite apt in providing a platform to scholars and participants to highlight and share the latest development in Data Acquisition, Data Transfer, Data Processing & Data Management.

I congratulate School of Computer Sciences in providing this crucial platform to researcher & scholars. I sincerely hope that this would enable them to adumbrate innovative and more effective strategies in the areas of Data Acquisition, Data Transfer, Data processing and Data Management including their role in every day’s life.

I sincerely hope that this international conference opens the doors for more such events in order to better understand and implement the practical aspects of the Data Acquisition Transfer, Processing & Management in the coming years.
Prof. Nabin K. Kole  
Pro Vice Chancellor,  
Lingaya's University

Message

I am extremely happy to learn that the School of Computer Science & Technology of Lingaya's University is organizing an International Conference on “Data Acquisition, Transfer, Processing and Management” on 28-29 March, 2014.

Computer Technology and IT Devices are developing at such a rapid speed that technological revolution will go to unpredictable stage. The Computer Technology has passed through several generations of developments starting from 1st generation with Vacuum Tubes and Magnetic Drums, up to 5th generation with Artificial Intelligence. Another generation of development known as 6th generation, is also becoming a practical reality.

Data Collection, Data Acquisition, Data Storage, Analysis and Management of the same, are becoming R&D activities of not only various research organizations but also of various corporate houses. Data are exploding in volume and variety, all around us. Several organizations are taking advantage of available data, through Warehousing and Analysis. Data Security, Privacy and Data Life Cycle Governance are also receiving priority among corporate and other researchers to a large scale.

Taking into account, the present day need and requirements for future developments, the School of Computer Science & Technology of Lingaya’s University, has properly selected an appropriate topic for this International Conference, to create a platform for discussion in global context.

My best wishes for a great success of the same.
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PREFACE

The advances in the area of Information Technology have made an impact in everybody's life, irrespective of the cast, creed, financial or social states of the people. A great transformation has taken place in this area in recent years. Starting from the applications in niche areas, where computers required a huge, isolated & cooled space, the computing devices have invaded in everybody's life. Now we talk of pervasive computing, Internet of Things, cloud computing and other related issues. All these areas include the capturing of the data in correct and timely manner; transferring the data in efficient, timely & secured manner, processing of the data in the most efficient manner; and finally managing the data in secured, efficient and easily accessible manner.

The data acquisition, transfer, processing and management happens in almost all the areas of our day to day life. To name a few areas are Image processing, sensor networks, geophysical explorations, Biology & Medicines, Lab measurements and analysis, Industrial Processes, Environmental monitoring & weather forecasting, defence & security, entertainment industry. In fact rapid developments in these areas have resulted in more intelligent, sensitive, secured and accurate methods of data acquisition, transfer, processing & management.

The developments in these areas have reached to a level that there is not even a single organisation, which provides products or solutions related to all the four areas viz. Data acquisition, transfer, processing and management. Even researchers throughout the world have been working either of these areas. A huge data exists related to progress made in each of these areas, throughout the world. This requires proper segregation, efficient storage and means of easy access to this information in a secured manner.

In the above back drop, the name of the conference ‘ICDATPM' was decided. This conference is intended to compile the strides made in each of these areas and provide a common platform to technocrats, researchers and users, for information sharing. In particular, officials working in the areas of mechanical engineering, electronics engineering, communication engineering, computer sciences and Information technology should find this conference extremely useful.

Editor
Brijesh Kumar
ACKNOWLEDGEMENTS

The editor wishes to express his appreciation and sincere gratitude to Dr. Picheswar Gadde, CEO, Linagya's University, Prof. M.A. Siddiqui, Chancellor Lingaya's University, Prof. R.K. Chauhan, Vice Chancellor Lingaya's University, Prof. Nabin K. Kole pro vice chancellor Lingyaa's University and Mr. Rajan Mital COE, Lingaya's University for their constant encouragement and guidance in Organizing committee of ICDATPM-2014.

Many thanks are to Prof. K.K. Aggarwal, Prof. Rajnish Prakash, Prof. K.D. Sharma, Prof. B.L. Raina and Prof. P.K. Suri, Kurukshetra for being the consistent source of inspiration from whom a lot has been learnt. They have been a motivating force and guide to me. I like to acknowledge the support of Prof. M.N. Hoda, Director BVICM, New Delhi; Mr. R.K. Vyas, Delhi University, New Delhi; Prof. Sunil Kumar Khatri, AIIT, Amity University Noida; Prof. Deepak Garg, Thapar University; Prof. Sunil Kumar Pandey, ITS Ghaziabad, Prof. Manoj Wadhwa, Echelon Institute of Technology; Dr. Namrata Aggarwal, NIFM, Faridabad; Dr. Ashwini Kush, Kurukshetra University;

I am also thankful to Dr. S.V.A.V. Prasad, Dean Corporate Affairs and Prof. A.K. Nadir, Dean Academics, for their consistent support.

I am also thankful to all the team members of ICDATPM organising committee at Lingaya's University, Faridabad, including faculty members viz Committee Members

Special thank goes to Dr. R.N. Malviya, Dr. Anubhav Kumar and Ms. Poonam Tanwar for helping in editing process and providing consistent support in making this conference a success.

The process remains incomplete, if I do not mention about my close associate, Dr. Tapas Kumar, Who helped in all possible manners in making this event a success.

Editor
Brijesh Kumar
Contents

Messages i-vi
  Dr. Picheswar Gadde, CEO
  Prof.R.K.Chauhan, Vice Chancellor
  Prof. Nabin K.Kole, Pro Vice Chancellor

Conference Committee vii

Preface viii

Acknowledgement ix

Keynote Address
• Importance of Data Acquisition, Transfer, Processing and Management in today’s context
  Prof. G.G.Senaratne

Invited Talks
• Use of Microwaves in breast cancer detection
  Prof. G.G.Senaratne
• Node Aggregation Policies in Wireless Sensor Networks
  Prof. Reena Dadhich
• Importance of Reliability in Data Acquisition, Transfer, Processing and Management
  Prof. Sunil Kumar Khatri
• Free and Open Source Softwares in Engineering Applications
  Prof. Brijesh Kumar

Papers

Data Acquisition

<table>
<thead>
<tr>
<th>Title(Paper-id), Author</th>
<th>Page</th>
</tr>
</thead>
<tbody>
<tr>
<td>4. ATM Security by Using Fingerprint Recognition (ICDATPM-1123) Ms. Nitasha Soni, Ms. Shilpa, Ms. Noopur</td>
<td>22</td>
</tr>
<tr>
<td>5. Mbed Microcontroller Based Temperature Measurement Using 1N4148 Diode sensor (ICDATPM-1155) K Jagadeesh, G Kiran, and Jatoth Ravi kumar</td>
<td>25</td>
</tr>
</tbody>
</table>
## Data Transfer

1. **Secure Remote Network Monitoring System** *(ICDATPM-1105)*
   Neeraj Rani, Kiran Kumar & Praveen Gupta  
   32

2. **Design and Development of Motorized Linear Translation Stage Using Microcontroller** *(ICDATPM-1113)*
   Gaurav Chhabra, Manpreet Singh and Usha Kumari  
   37

3. **I P T V – Triple Play Service** *(ICDATPM-1120)*
   Khushboo Yadav and Kamlesh Sharma  
   45

4. **Detailed Review of Wavelet Transform Approach for P300 Based BCI Systems** *(ICDATPM-1130)*
   Mandeep Kaur  
   52

5. **A Review of Routing Protocols Of Manet** *(ICDATPM-1149)*
   Ms. Shilpa Dixit, Nitasha Soni and Noopur Relan  
   59

6. **Internet Protocol Version 6(Ipv6)** *(ICDATPM-1152)*
   Sandeep Yadav and Sharanya Chandran  
   66

7. **Simulation Tools for Mobile Ad-hoc Network** *(ICDATPM-1163)*
   Renu Sheoran and Ritu Sheoran  
   70

8. **WSN Clustering Protocol** *(ICDATPM-1178)*
   Ms. Meenakshi, Ms. Vasudha vashisht and Mr. Amit Chugh  
   77

9. **Provide security b/w Sender & Receiver in Bluetooth Using Java** *(ICDATPM-1183)*
   MS. Ruchi Sharma, Savita Bainsla and Mr. Yogender Sharma  
   84

    Meenakshi Sharma and Manoj Kumar Jain  
    90

11. **A Comprehensive Survey on WiMAX Scheduling Approaches** *(ICDATPM-1188)*
    Neha Kak  
    97

## Data Processing

1. **Enhancing Authentication on RFID Tags Using CA** *(ICDATPM-1101)*
   Girija R and Sathish R  
   105

2. **Prediction of Next Search Query Using Association Rule Mining** *(ICDATPM-1107)*
   Rosy Madaan, Manvi Breja, A.K. Sharma and Ashutosh Dixit  
   111

3. **Summarization of Search Results Based On Concept Segmentation** *(ICDATPM-1109)*
   Naresh Kumar, Rajender Nath, Sudhir Kumar, Amit Kumar, Manish Jain and Kaushal Kumar Dubey  
   116

4. **Sentiment Analysis based Approach for Interlinking Events to Analyze the Performance of Government during Disaster** *(ICDATPM-1110)*
   Kanika Saini, Rakesh Ranjan, Rupesh Kumar Mishra  
   123

5. **Location Based Search Privacy Mechanism for Mobile Social Network** *(ICDATPM-1115)*
   Vinay Kumar, Dr. Pradeep Tomar and Dr. Karan Singh  
   130
1. **A Novel method for Image Segmentation & Comparative Study** *(ICDATPM-1117)*
   Poonam Tanwar and Sheetal Rishi  
   135

2. **Performance Comparison of Instrumentation Amplifiers – A Beginner’s View** *(ICDATPM-1119)*
   Bhargava Peddiraju, Ravi Kumar Jatoth and Nagaraj Duggirala  
   144

3. **A Weighted Relevance Response Approach for Web Image Retrieval Using Textual and Visual Features** *(ICDATPM-1122)*
   N. Mohana Eswari  
   151

   Ravindra Kumar Chahar and Dr. S.V.A.V. Prasad  
   158

5. **Semantic Clone Detection for UML Based Software Systems** *(ICDATPM-1129)*
   Sahil Sethi, Parveen Gorya  
   165

6. **Analysis and Design of a Bidirectional DC–DC Converter for Super Capacitor** *(ICDATPM-1132)*
   Poonam Mavi, Dr. Ashok Arora and Dr. Pradeep Dimri  
   172

7. **Designing of an Isolated Full bridge dc-dc Converter** *(ICDATPM-1133)*
   Poonam Mavi, Dr. Ashok Arora and Pradeep Dimri  
   178

8. **A Novel Isolated Bi-directional DC-DC Converter for Renewable Energy Storage Systems** *(ICDATPM-1141)*
   Poonam Mavi, Dr. Ashok Arora and Dr. Pradeep Dimri  
   184

9. **A Novel Design of a Bi-Directional DC-DC Converter for Egenerative Energy Storage System** *(ICDATPM-1144)*
   Poonam Mavi, Dr. Ashok Arora and Dr. Pradeep Dimri  
   189

10. **A Proposed Approach on Edge Detection** *(ICDATPM-1145)*
    Palvi Rani, Poonam Tanwar,  
    197

    Dolly Sharma, Shailendra Singh and Trilok Chand  
    206

12. **Semi-hidden Target Recognition in Gated Viewer Images Fused with Thermal IR Images** *(ICDATPM-1159)*
    Kalpana Jaswal, Seema Rawat and Praveen Kumar  
    209

    Ms Monika and Dr. Tapas Kumar  
    217

14. **Inverse Computation of Microwave Scattering Signals for Foreign Object Identification Such as Breast Screening** *(ICDATPM-1175)*
    G. G. Senaratne  
    221

15. **Predicting Success of Nursery Schools Using Artificial Neural Networks** *(ICDATPM-1176)*
    Manoj Kumar Jain and Suman Kumar Jha  
    229

16. **Video Image Processing: Detection and Tracking Of Color** *(ICDATPM-1180)*
    Kshiti Wahi, Harpreet Kaur, Utsav Singh and Kamlesh Sharma  
    234

17. **Performance Evaluation of Various Classification Techniques by Using Educational Dataset** *(ICDATPM-1189)*
    Dr. Anubhav Kumar, Dr. Arvind Kumar Sharma and Ms. Anuradha  
    239
23. String Algorithms for Counting DNA Nucleotides, Transcribing DNA to RNA and Complementing a Strand of DNA (ICDATPM92)
Pardeep Kumar, Dolly Sharma

Data Management

1. Leveraging Artificial Intelligence Benefits in Control Engineering (ICDATPM-1103)
Snigdha Chawla, Amit Chugh, and Nitasha Soni

2. Multicast Model Based on Grouping for Access Control in Online Social Networks (ICDATPM-1116)
Deepa Bharti, Sandhya Tarar and Dr. Karan Singh

3. Optimization of Power Consumption in VLSI Circuit (ICDATPM-1118)
Praveen Kumar Gupta and Jagdeep Kaliraman

4. Review: Cache-timing Attacks on AES (ICDATPM-1125)
Priyanka Mittal and Poonam Tanwar

5. Cloud Computing a Fundamental Change in Computer Architecture (ICDATPM-1128)
Kamlesh Sharma, Lalit Sharma, Khoosbu Yadav and Dr. S.V.A.V Prasad

Ruchi Dagar and Vidushi Rawal

Annu Dangi and Parveen Gorya

8. Review: Log Mining in Databases (ICDATPM-1135)
Romika Bhatia and Latha Bandha

Vidusshi Rawal and Ruchi Rani

10. Service Quality of Metro Railways in Delhi - A Brief Study of Delhi and NCR (ICDATPM-1150)
Pinki Singh

Robin Gaur and Parveen Gorya

12. Comparative Study of Educational ERP System (ICDATPM-1168)
Suman kumar Jha, Manoj Kumar Jain Jain and Md Shamsuddin

13. Ls Retail- Ms Dynamics Navision- Inventory Setup And Management (ICDATPM-1172)
Divya Sarup Sharma and Dr. Anubhav Kumar

Harshit Chauhan, Rina Ramjibhai Patel and Vandana Gupta

15. Review of Composite Helical Spring Analysis with Ansys Product Software (ICDATPM-1182)
S.K Bhardwaj, Lakhwinder Singh and M.L. Aggarwal
Data Acquisition
Transient Stability Analysis of Standard IEEE39 Bus System Using Mi-Power Software Simulation

Aditya Narula, Sheila Mahapatra, Amandeep Singh Rattan
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Abstract—Transient stability of a power system is the ability of the system to return to its normal operating condition once the disturbance is cleared from the power system. The disturbances subjected to the system may be occurrence of fault, sudden change in load data etc. The concept of transient stability plays a significant role in maintaining the stability of a multi-machine power system. This paper presents load flow and transient stability analysis of a standard IEEE 39 bus 10 generator system using MiPower software simulation. The system is subjected to large disturbances which include three phase to ground fault, single line to ground fault, generator outage and sudden change in load. Results for change in frequency, terminal voltage, current and swing curve are realized.

Keywords: Standard IEEE 39 bus system, MiPower software, three phase fault, single line to ground fault, generator outage, sudden change in load

I. INTRODUCTION

In recent times power system stability has become one of the most important aspects in the field of power system operation and control. The system may be subjected to a major disturbance like a three phase fault, generator outage etc. and it is important to keep the frequency and voltage variations within the prescribed limits and most importantly keeping the system in synchronism.

The disturbances subjected to the system may be small or large. Small disturbances in a power system may be due to frequent switching of appliances or by small change in load data. The magnitude of these disturbances is very small compared to the system. Thus these disturbances may be neglected.

Large disturbances in a system may be due to loss of transmission line, lightning strike, fault on a line or loss of excitation. These disturbances in worst conditions may lead to the power system losing its synchronism. Thus these disturbances cannot be neglected.

The synchronous stability or rotor angle stability can be classified into

- Steady State Stability
- Transient State Stability

The Steady State Stability of the system deals with the ability of the system to remain stable under gradual or slow disturbances.

The Transient State Stability deals with the ability of the system to transfer maximum power after being subjected to large and sudden disturbances without losing its synchronism.

A. Load Flow Analysis:

Load flow analysis or power flow analysis is a steady state analysis. It is used to analyze four major components at a particular bus of a power system i.e. its voltage, load angle, real power and reactive power.

Based on these four components the buses in a power system can be classified as follows

<table>
<thead>
<tr>
<th>Bus Type</th>
<th>Known Component</th>
<th>Unknown Component</th>
</tr>
</thead>
<tbody>
<tr>
<td>Load Bus or PQ Bus</td>
<td>P and Q</td>
<td>δ and</td>
</tr>
<tr>
<td>Generator Bus or PV Bus</td>
<td>P and</td>
<td>V</td>
</tr>
<tr>
<td>Slack Bus or Reference Bus</td>
<td>δ and</td>
<td>V</td>
</tr>
</tbody>
</table>
B. Static or steady state Power Flow Equations

For any bus i current injections are
\[ I_i = \sum_{k=1}^{n} Y_{ik} V_k \]  \hspace{2cm} (1)
\[ I_i = \sum_{k=1}^{n} |Y_{ik}| V_k \angle \theta_{ik} \delta_k \]  \hspace{2cm} (2)

We know
\[ S_i = V_i I_i^* = |V_i| \angle \delta_i I_i^* \]  \hspace{2cm} (3)
\[ S_i^* = V_i^* I_i = |V_i| \angle \delta_i I_i \]  \hspace{2cm} (4)

Thus Complex powers at bus i

- \[ P_i = \sum_{k=1}^{n} |Y_{ik}| V_k V_i \cos (\theta_{ik} + \delta_k - \delta_i) \]  \hspace{2cm} (5)
- \[ Q_i = \sum_{k=1}^{n} |Y_{ik}| V_k V_i \sin (\theta_{ik} + \delta_k - \delta_i) \]  \hspace{2cm} (6)

Where
\[ S = \text{apparent power} \]
\[ P = \text{active power} \]
\[ Q = \text{reactive power} \]
\[ I = \text{current} \]
\[ V = \text{voltage} \]
\[ \delta = \text{power angle} \]

In this paper load flow analysis is performed using fast decoupled method with a multiplying factor of 1. Load flow results are shown in the TABLE II.

<table>
<thead>
<tr>
<th>TABLE II. LOAD FLOW ANALYSIS</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Load Flow Analysis</strong></td>
</tr>
<tr>
<td>Total Generation: 6174.538 (1537.730)</td>
</tr>
<tr>
<td>Total Load: 6097.100 (1409.100)</td>
</tr>
<tr>
<td>Total Losses: 77.449790 (128.641463)</td>
</tr>
</tbody>
</table>

TABLE II depicts that the test system is balanced under normal operating conditions.

C. Transient State Stability

Transient stability is the ability of the power system to transmit power and remain stable or in step after being subjected to a major disturbance.

We assume a synchronous machine connected to infinite bus as shown in “Fig. 1”.

\[ S = \frac{|E|}{X} \sin \delta + j \left( \frac{|E|}{X} \cos \delta \right) \frac{V_i}{X} \]  \hspace{2cm} (7)
\[ S = \frac{|E|}{X} \angle (90^\circ - \delta) \frac{V_i^2}{X} \]  \hspace{2cm} (8)

The real power delivered to system is
\[ P_e = \text{Re} [S] = \frac{|E|}{X} \sin \delta = P_m \sin \delta \]  \hspace{2cm} (11)

Where
\[ P_m = \frac{|E|}{X} \]  \hspace{2cm} (12)

Where
\[ S = \text{apparent power} \]
\[ E = \text{sending end voltage} \]
\[ V = \text{Voltage at infinite bus} \]
\[ \delta = \text{power angle} \]
\[ X_g = \text{generator reactance} \]
\[ X_L = \text{transmission line reactance} \]

Transient stability is analyzed using Swing Equation; it gives the variation in rotor angle (with stator as reference) over the period of time.

\[ M \frac{d^2 \delta}{dt^2} = P_m - P_e = P_2 \text{ in Megawatt} \]  \hspace{2cm} (13)
\[ H \frac{d^2 \delta}{dt^2} = P_m - P_e = P_4 \text{ in per unit} \]  \hspace{2cm} (14)
Where
\[ M = \text{angular momentum} \]
\[ H = \text{moment of inertia} \]
\[ f = \text{frequency} \]
\[ P_m = \text{mechanical power input} \]
\[ P_e = \text{electrical power output} \]
\[ P_a = \text{accelerating power} = P_m - P_e \]

II. SYSTEM MODEL

The network under consideration is a standard IEEE 39 bus system or the 39 New England system as shown in “Fig. 2”. The system comprises of 39 buses and 10 generators. The total load present on the system is 6097.1 Mega Watt and 1409.1 Mega Vars. The system comprises of 12 transformers which act as isolators. Bus number 39 is taken as slack bus for the test circuit.

The base Mega Volt Amperes taken for the test system is 100 Mega Volt Amperes and the base frequency taken for the test circuit is 60 Hertz.

III. SIMULATION AND RESULTS

The load flow analysis of the standard IEEE 39 bus system is shown in “Fig. 3”.

Fig. 2. IEEE standard 39 bus system

A. Transient analysis for Three Phase to Ground Fault

The rarest and the severest fault that occurs in a power system is a three phase symmetrical fault. The system stability can be best checked after executing a three phase to ground fault. The system is subjected to a three phase to ground fault. The fault occurs at bus 39 at \( t=0.1 \) seconds and is cleared at \( t=0.2 \) seconds. Thus the bus is subjected to fault for 0.1 seconds.

Fig. 1. Current curves for 10 generators 39 bus system

Fig. 3. Load Flow Analysis Simulation in MiPower
The magnitude of fault current at bus 39 is 43.3602 kilo Amperes.

The maximum and minimum variation in frequency is 60.26 Hertz and 59.95 Hertz respectively.

The 10 generators remain in synchronism and are able to return to a condition in proximity to the initial condition.

The terminal voltage of generator 10 reduces to 0 per unit during fault.

When a three phase to ground fault occurs in the system the system remains stable due to continuous change of the rotor angles. The frequency increases above the limits and the terminal voltage decreases below the prescribed limits. The current rises continuously.

B. Transient analysis for Single Line to Ground Fault

Single line to ground fault is the most frequent fault and the least severe of all. The system is subjected to a single to ground fault. The fault occurs at bus 39 at t=0.1 seconds and is cleared at t=0.2 seconds. Thus the bus is subjected to fault for 0.1 seconds. The fault impedance is taken 0.
The magnitude of fault current at bus 39 is 19.594 kilo Amperes.

The maximum and minimum variation in frequency is 60.13 Hertz and 59.99 Hertz respectively.

The 10 generators remain in synchronism and are able to return to a condition in proximity to the initial condition.

The terminal voltage of generator 1 reduces to 0.56 per unit during fault.

When single line to ground fault occurs in the system the system remains stable due to continuous change of the rotor angles. The frequency deviates and the terminal voltage drops below the prescribed limits. The current rises continuously.

C. Transient analysis for Generator Outage

Generator Outage means to turn off the generator as and when required for maintenance but the following points need to be addressed before moving forward. To execute the above, the Generation Company should plan as Generator Outage Plans. In this case we remove the generator that is generator 4 connected at bus 33 at t=0.2 seconds.

The frequency drops from 60 Hertz rapidly after the load data is changed at t=0.5 seconds.

Generator 4 connected to bus 33 loses synchronism and is unstable while the remaining generators are stable.

The terminal voltage of generator 4 reduces to 0.85 per unit at t=0.2 seconds.

When the generator outage occurs the frequency dwindles at high rate. The terminal voltage is within limits but the current increases.
D. Transient analysis for Change in Load data

Sudden change in load affects the system adversely. Simulating the similar situation at bus 27 of the test circuit, the load data of load 5 is changed from $281+75.5j$ to $550+100j$ at $t=0.5$ seconds.

![Current curve for 10 generators 39 bus system](image1)

![Frequency curve for 10 generators 39 bus system](image2)

![Swing curves for 10 generators 39 bus system](image3)

Fig. 19. Terminal Voltage curve for 10 generators 39 bus system

- The frequency drops from 60 Hertz rapidly after the load data is changed at $t=0.5$ seconds.
- The 10 generators remain in synchronism and are able to return to a condition in proximity to the initial condition.
- The terminal voltage reduces to a minimum of 0.96 per unit after the load data is changed.

When the parameters of load are changed, the system is stable but the rotor angle of the bus where the load is connected becomes negative. The frequency decreases but the terminal voltage is within specified limits. The current of the bus at which load is connected increases.

CONCLUSION

This paper gives the simulated result for the maximum frequency deviation, deviation of terminal voltage, deviation in current and swing curve response for different types of disturbances subjected to the test system. The simulated result depicts the variation of the above parameters for all the generators simultaneously and it is observed that the system retains its stability. This shows that the transient simulation studies done is suitable for the given test system and it can be further implemented to large interconnected power system. The stability studies conducted on the above test circuit will be helpful for a power engineer to determine the condition for stability.

REFERENCES


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Abstract: The evaluation of software layered technology quality attribute significance depends on its layers implementation. This may be a difficult task and the Analytic Hierarchy Process seems to provide an effective approach for properly quantifying the pertinent data. Even though, there are many critical issues that a decision maker needs to be aware of. This paper examines some of the practical and computational issues involved when the AHP method is used in this software engineering applications to find out the quality significance.

Key words: Quality focus, methods, process, tools, AHP, Priorities, Alternatives, Pair wise comparisons, Priority Vector, Eigen vector.

1. INTRODUCTION

The nature and complexity drastically and significantly changed in the last 40 years. The computer applications run on a single processor, produced alphanumeric output and received their output from a liner source. In recent times software has become more complex and it is being executed on graphical user interface with client server technologies on multiprocessor systems through different operating systems in distributed environment.

The software engineering is concerned with the practices of developing and delivering the useful software. The software products may be developed for a particular customer or may be for general purpose. Software Engineering is an engineering discipline and its goal is to develop the cost effective software products with in the stipulated and budget.

The software product is intangible which consists of programs and associated documentation. The software engineering is systematic approach of software development which is concerned with all the aspects of software production.

The subject embedded concept of Project Planning, Project Tracking, Formal Inspections, Configuration Management, Software Quality Assurance, and Risk Management etc. It has contributed a great deal in its short span of time.

Fig. 1. Components of software product

This paper presents the significance of quality attributes in the software layered technology using Analytic Hierarchy Process. The decision making process depends on multiple parameters and criteria. The parameters are like reliability, usability, maintainability, portability, efficiency etc. The Section 2 explains software engineering process models and its over view. The section 3 states the various layers in Software Layered Technology. Section 4 describes the Analytic Hierarchy Process in evaluation of quality attribute significance in software layered technology with mathematical derivations. Finally a discussion about future scope and conclusions is given in the Section 5.
2. SOFTWARE PROCESS MODELS

The process is defined as the way of approach in which we produce the software product. It provides the framework which contains set of activities for software development. Increasing demand and competition in software industry demands for more hiring smart, knowledgeable, innovative developers and busing the latest development tools. The strengthen of the software based on the process model. Many organizations are looking at software process development as a way to improve the quality, productivity, predictability of their software development, and maintenance efforts. There are some of common activities of all the software processes.

a. To develop an informal software specification of the system.
b. To design & develop the software which meets specifications.
c. To validate the software to ensure the customer requirements.
d. To maintain & evaluate the software according to the process and technology change management.

Software processes are complex, intellectual and involves the participation of human judgment. The success of the software process model is based on the human judgment embedded with the suitable software tool.

There are good number of success stories, when it comes to development of software process models. However, some organizations are failing to achieve significant results in developing and implementing the software process models due to the following reasons.

a. Insufficient Time: Unrealistic time schedules leave insufficient time to do the project work. Customers and senior managers are demanding more software, of higher quality in short span of time.
b. Lack of Knowledge & Innovative skills: The next obstacle to the developers is unfamiliarity with industry best practices. Normally software developers do not spend much time in reading the subject literature related to best practices.
c. Lack of motivations: The CMM level organizations can’t motivate the sub systems in positive direction. The organizations launch process improvement initiatives for the wrong reasons.
d. Insufficient commitment and dedication:

The software process management fails, despite best of intentions, due to the lack of true commitment. It starts with a process assessment but fails to follow through with actual changes. Management sets no expectations from the development community around process improvement. Due to insufficient resources, no improvement plan, no road map, and motivation of new processes.

The software process model is representation of software process. Each process model represents particular perspective, it will provides partial information about that process. For many large and complex systems, of course, there is no single process that is use. Various processes are used to develop different parts of the system.

1. The water fall model is linear sequential model which starts with requirement specification, progresses with design, development and end with implementation and evolution.
2. Incremental model: The incremental model is representation contain the same phases of water fall model but in the successive increments.
3. Evolutionary development: This approach is set of interleaving the activates and specifications in development and validation. The system starts at initial stage with abstract specifications, then refined with customer input to produce new refinement in repetitive which satisfies the customer’s needs.

3. SOFTWARE LAYERED TECHNOLOGY

The software engineering is layered technology. It encompasses a process, the management, technical methods, and use of tools to develop the software products. The objective of any software engineering approach is committed for quality factor.

The various philosophies which are defined in Total Quality Management, Six Sigma, Statistical analytical processes are targeted software development towards improvement of quality culture.
The software layered technology as classified its activities based on importance as quality focus layer, process layer, methods layer and tools layer.

**Fig. 2.** Software Layered Technology

**Quality Focus layer** : The bedrock of software engineering is quality focus. The quality management is backbone of software layered technology which consists of Total Quality Management Tools, Six sigma methods etc. The software product quality should meet its specification. The software product should fulfill the customer quality requirements (i.e efficiency, reliability, etc), developer quality requirements (maintainability, reusability, etc), users (usability, efficiency etc). The quality constraints are non functional requirements. The some of quality requirements are difficult to specify in an unambiguous way. Software specifications are usually incomplete and often inconsistent.

**Process layer** : The process layer is foundation of software engineering. *process* defines a frame work for timely delivery of software. The key process areas form the basis for management control of software projects. The various tasks can be performed in this layer.

- Determining Deliverables
- Establishing milestones
- Software configuration / Change management.
- Software Quality Assurance

**Methods Layer** : Software engineering *methods* provide the technical knowledge (i.e “how to’s”) for building software. Methods comprises various array of tasks of the following.

- Requirement Analysis
- Design
- Program Construction.

- Testing and support.

**Tools layer** : The software Engineering *Tools* provide automated or semi-automated support for the process and methods. The tools are used to bring automation in software development process.

Ex : CASE (Computer Aided Software Engineering) and Rational Rose etc.

When the tools are integrated so that information created on tool can be used by another, a system for the support of software development called the Computer aided software Engineering. The CASE tools may also include editors, database, test case generators and code generator which automatically generates the source for the system models.

**Software Process Frame work**

The process framework consists of process activities which are suitable for all software projects irrespective of its size and complexity. The whole software process framework contains the umbrella activities which exists the set of framework activities embedded with software engineering actions. The each action is highlighted with individual work tasks that accomplish some part of the work implied by the action.

In general vast majority of software projects follow generic process framework as given below.

**Communication** : Transparent communication with the customer and other stakeholders about gathering of requirements.

**Planning** : Describes the technical tasks and establish the milestones, fix the work schedules and required resources for achieve the goal.

**Modeling** : Encompasses the representation of models for better understanding of S/W requirements. The modeling is pictorial representation of proposed solution in terms of DFD’s, Grid Charts, Flowcharts, Decision trees and Decision tables etc.

**Construction** : Combine the development, debugging and testing process for maintaining the quality of software product.

**Deployment & Evolution** : The deployment is implementation of software product at the real time environment. Evolution is a collection of feedback
from the stake holders and maintain the software product for customer satisfaction.

The generic view of framework describes the no. of umbrella activities typically as

**Risk Management**: Assesses and estimates risks that may affect the outcome of the project or the quality.

**Software Quality Assurance**: Verify the activities to ensure the software quality focus.

**Software project tracking and control**: Examine the project progress with plan, milestones and work schedules.

**Formal Technical Review**: Remove the errors in design and code generation before going to next phase.

**Configuration management**: Manages the effect of change in terms of process and technology throughout the S/W process.

**Measurement**: The measurement is activity to ensure the product which meets the customer needs.

**Work products**: Creates the work projects like models, documents, forms, lists etc.

4. **ANALYTIC HEARARCHY PROCESS**

The Decision making on the basis of several criteria and alternatives is very difficult process. We need a decision method that enables a quantitative comparison between layers based on the quality attributes in software layered technology. Such problem solved with the Analytic Hierarchy Process (AHP). The Analytic Hierarchy Process invented by the **Saaty** in 1980 and improved by Vargas in 2001. The AHP was used in multi-criteria decision making and management science by Anderson et al., in 2000. It is a powerful and flexible tool for decision-making in complex multi criteria problems. The solutions can be both objective and subjective. This tool is developed to solve the various issues and derive the solutions.

In this paper the attention is focused on the comparative significance of quality attributes in software layered technology using AHP decision making method.

**Structure of AHP method**

Analytic hierarchy process is a expert mathematical model which divides the main problem into smaller and more detailed elements.

Decision by AHP method can be divided into three different levels

1. **hierarchy**, 
2. **priorities**, 
3. **consistency**

Designing a structured AHP hierarchy means developing a system consisting of a goal of decision making process.

**Priorities**

After sorting their own set of criteria and the establishment of a hierarchical structure at all levels of assessment, various alternatives or criteria that affect the assessment through verbal explanations and figures are compared. The result is given by the weight in proportion to the scale of alternatives and criterions.

**Weight allocation**

The correct and responsible determination of the individual sub-scales of assessment criteria is one of the key tasks in solving multi criteria problems. It is therefore necessary to know the solved issue well and know the importance and impact of the criteria used to evaluate the result achieved.

This method allows to gather knowledge about a particular problem, to quantify subjective opinions and to force of alternative in relation to established criteria.

1. Define the problem and the main objectives to make the decision.
2. Build a hierarchical structure as Figure 3, the root node is the objective of the problem, Intermediate level as criteria’s and lower levels contain the alternatives. The entire structure overviews the criteria and the alternatives.
3. Construct a set of pair wise comparison matrices.

The element in an upper level is used to compare the elements in the level immediately below with respect to it. For each comparison matrix, find the Eigen value, consistency index CI, consistency ratio
CR, and normalized values for each criteria / alternative.  

4. Use the priorities obtained from pair wise matrix in global matrix. The scale for rating characteristics should be established and described in a precise way. Do this every element. Then for each element in the level below add its weighted values and obtain its overall or global priority. Continue this process of weighting and adding until the final priorities of the alternatives in the bottom most is obtained. The final value is used to make a decision about the objective.

**Mathematical Example:**  
The following example is to find the weighted significance of quality attributes in quality focus layer can be evaluated in terms of decision criteria of remaining layers ie. Process, Methods, Tools of the software layered technology. The pair wise comparison matrix represent the corresponding judgment on scale of relative importance.

**Table 1.** Scale of Relative importance (according to the Saaty (1980))

<table>
<thead>
<tr>
<th>Weight</th>
<th>Definition</th>
<th>Explanation</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Equal importance</td>
<td>Two activities in equal importance</td>
</tr>
<tr>
<td>3</td>
<td>Moderate importance</td>
<td>One activity moderate over another</td>
</tr>
<tr>
<td>5</td>
<td>Strong importance</td>
<td>One activity strong over another</td>
</tr>
<tr>
<td>7</td>
<td>Very strong importance</td>
<td>One activity very strong in practice over another</td>
</tr>
<tr>
<td>9</td>
<td>Extreme importance</td>
<td>One activity Extreme over another</td>
</tr>
<tr>
<td>2,4,6,8</td>
<td>Intermediate values between two activities</td>
<td>When compromise is needed.</td>
</tr>
<tr>
<td></td>
<td>Reciprocals of above non Zero</td>
<td>If activity I has of above non nonzero numbers assigned to it when compared with activity j, then j has the reciprocal value when compared with it</td>
</tr>
</tbody>
</table>

The next step in pair wise comparisons, the corresponding maximum left eigenvector is approximated by using geometric means of each row. An evaluation of the eigenvalue method can found in (Triantaphyllou and Mann, 1990). Initially the consistency index (CI) can be estimated. This is done by sum of columns in the judgment matrix and multiply the resulting vector by the vector of priorities (i.e approximated eigenvector) obtained earlier. This result the approximation of the maximum eigenvalue. denoted by λmax. Then, the CI value measured by using the formula as $CI = \frac{\lambda_{max} - n}{n - 1}$. Then after the consistency ratio CR is obtained by dividing the CI value by Random Consistency Index (RCI) as the table given below.

**Table 2.** Random Consistency Index based on matrix size (adopted from Saaty, 2000)

<table>
<thead>
<tr>
<th>Matrix Size (n)</th>
<th>Random Consistency Index</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>2</td>
<td>0</td>
</tr>
<tr>
<td>3</td>
<td>0.58</td>
</tr>
<tr>
<td>4</td>
<td>0.90</td>
</tr>
<tr>
<td>5</td>
<td>1.12</td>
</tr>
<tr>
<td>6</td>
<td>1.24</td>
</tr>
<tr>
<td>7</td>
<td>1.32</td>
</tr>
<tr>
<td>8</td>
<td>1.41</td>
</tr>
<tr>
<td>9</td>
<td>1.45</td>
</tr>
</tbody>
</table>

The weights of importance of the criteria are also determined by using by using pair wise comparisons. If the problem has M alternatives and N criteria, then the decision maker is required to construct N judgment matrices (each criteria) of order M*M and one judgment matrix of order N*N ( for N criteria) . Finally, the decision matrix the final priorities denoted as $A^i_{AHP}$.

$$A^i_{AHP} = \sum_{j=1}^{N} A_{ij} w_j, \text{ for } i = 1, 2 \ldots M ----- (1)$$

Suppose three quality attributes i.e Portability (P), Reliability (R), Maintainability (M) significance can be evaluated on based of its quality focus, process,
methods and tools in the pair wise comparisons and AHP methodology.

The figure 3 shows the hierarchical decomposition of criteria, sub criteria, and alternatives. The Level 0 shows the overall goals of “significance of quality attributes”. The next level, namely level 1 shows the criteria of various levels of software layered technology. Its next level namely level 2 is the highest level shows the quality attributes as alternatives.

![Hierarchical decomposition of Criteria’s Criteria’s & alternatives](image)

The weights of alternatives with respect to each of the criteria mentioned in the tables 3 to 5 and the its priority vectors represented in bar graphs from figures 4 to 6.

The first table is with respect to the process and ranks of the three quality attributes as follows

**Criteria [ C1 ] :** Process

<table>
<thead>
<tr>
<th>PROCESS</th>
<th>P</th>
<th>R</th>
<th>M</th>
<th>Priority Vector</th>
</tr>
</thead>
<tbody>
<tr>
<td>P</td>
<td>1</td>
<td>5</td>
<td>9</td>
<td>0.748</td>
</tr>
<tr>
<td>R</td>
<td>1/5</td>
<td>1</td>
<td>3</td>
<td>0.180</td>
</tr>
<tr>
<td>M</td>
<td>1/9</td>
<td>1/3</td>
<td>1</td>
<td>0.072</td>
</tr>
<tr>
<td><strong>Total Priority</strong></td>
<td></td>
<td></td>
<td></td>
<td>1</td>
</tr>
</tbody>
</table>

$\lambda_{max} = 3.052, \quad CI = 0.026, \quad CR = 0.045$

Table 3. Weights of Alternative w.r.t Process

**Fig. 4.** Weights of alternatives w.r.t Process

The next two matrices are respectively judgments of the relative merits of portability (P), reliability (R), maintainability (M) with respect to methods and tools of software layered technology.

**Criteria [ C2 ] :** Methods

<table>
<thead>
<tr>
<th>Methods</th>
<th>P</th>
<th>R</th>
<th>M</th>
<th>Priority Vector</th>
</tr>
</thead>
<tbody>
<tr>
<td>P</td>
<td>1</td>
<td>7</td>
<td>1/5</td>
<td>0.250</td>
</tr>
<tr>
<td>R</td>
<td>1/7</td>
<td>1</td>
<td>1/8</td>
<td>0.060</td>
</tr>
<tr>
<td>M</td>
<td>5</td>
<td>8</td>
<td>1</td>
<td>0.690</td>
</tr>
</tbody>
</table>

**Total Priority**

$\lambda_{max} = 3.412, \quad CI = 0.206, \quad CR = 0.356$

Table 4. Weights of Alternative w.r.t Methods

**Fig. 5.** Weights of alternatives w.r.t Methods
Criteria [ C3 ] : Tools

<table>
<thead>
<tr>
<th>Methods</th>
<th>P</th>
<th>R</th>
<th>M</th>
<th>Priority Vector</th>
</tr>
</thead>
<tbody>
<tr>
<td>P</td>
<td>1</td>
<td>5</td>
<td>4</td>
<td>0.665</td>
</tr>
<tr>
<td>R</td>
<td>1/5</td>
<td>1</td>
<td>1/3</td>
<td>0.104</td>
</tr>
<tr>
<td>M</td>
<td>1/4</td>
<td>3</td>
<td>1</td>
<td>0.231</td>
</tr>
</tbody>
</table>

Total Priority = 1

$\lambda_{max} = 3.131, \quad CI = 0.066, \quad CR = 0.113$

Table 5. Weights of Alternative w.r.t Tools

Fig. 6. Weights of alternatives w.r.t Tools

The final step describes the judgment matrix table.6 based on the criteria importance of the three layers of software layered technology.

<table>
<thead>
<tr>
<th>3- CRITERIA</th>
<th>P</th>
<th>R</th>
<th>M</th>
<th>Priority Vector</th>
</tr>
</thead>
<tbody>
<tr>
<td>P</td>
<td>1</td>
<td>5</td>
<td>7</td>
<td>0.724</td>
</tr>
<tr>
<td>R</td>
<td>1/5</td>
<td>1</td>
<td>3</td>
<td>0.193</td>
</tr>
<tr>
<td>M</td>
<td>1/7</td>
<td>1/3</td>
<td>1</td>
<td>0.083</td>
</tr>
</tbody>
</table>

Total Priority = 1

$\lambda_{max} = 3.111, \quad CI = 0.056, \quad CR = 0.096$

Table 6. Weights of Criteria’s (layers)

Figure 7 shows the weights of layers process, methods and Tools layers represented in bar graphs.

Fig. 7. Weights of Criteria’s (Layers)

The previous priority vectors are used to form the entries of the decision matrix for this problem. The decision matrix and the resulted final priorities (as calculated according to formula(1) ) as follows.

<table>
<thead>
<tr>
<th>Quality Focus</th>
<th>C1</th>
<th>C2</th>
<th>C3</th>
<th>Quality Significance</th>
</tr>
</thead>
<tbody>
<tr>
<td>PORT [P]</td>
<td>0.541</td>
<td>0.048</td>
<td>0.055</td>
<td>0.645</td>
</tr>
<tr>
<td>RELIA [R]</td>
<td>0.131</td>
<td>0.012</td>
<td>0.009</td>
<td>0.151</td>
</tr>
<tr>
<td>MAINT [M]</td>
<td>0.052</td>
<td>0.133</td>
<td>0.019</td>
<td>0.204</td>
</tr>
</tbody>
</table>

Total Priority = 1

Table 7. Significance of the Quality attributes

The significance of the attributes in quality focus layer shown in the figure. 8 with bar graphs is based on performance of remaining layers.

Fig. 8. Significance attributes in Quality Focus of Software Layered Technology.
Therefore, the quality significance of the Portability is followed by Maintainability which is followed by Reliability.

5. CONCLUSIONS AND DISCUSSION

The AHP provides a convenient approach for solving complex Multi Criteria Decision Making problems in software engineering. The Expert Choice (1990) software, which significantly contributed to wide acceptance of AHP methodology. The numerical example in this paper, along with extensive research of authors suggest that when some alternatives to be very close to other, then the decision maker needs to be very cautious. The MCDM method may never end, research in this area of decision making is still critical and vary valuable in many scientific and software engineering applications.

6. REFERENCES

The Role Of Knowledge Information Systems And Its Effectiveness In The Decision Making: An Information Systems Perspective

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Abstract: Nowadays information systems play a very important role in improving an organisation’s performance and its increased competitive capacity. Therefore, it is essential for organizations to decide what are most important business processes and core competencies that have to be supported by an information system and what kind of information system has to be implemented for the organisation’s requirements. In this context, this paper firstly speaks about understanding of knowledge information system and primary steps in decision making process. The second part shows the functions of the knowledge information systems in the organization in different perspectives. The third part refers to key objectives of the knowledge information system and fourth part overviews the typical knowledge retrieval system and last part analyses the role of information systems in the organization for its effectiveness in decision making process.

Keywords: Knowledge Management Systems, KMS, knowledge networks, Knowledge discovery, Explicit knowledge, Tacit knowledge, Artificial intelligence, Knowledge repository.

I. INTRODUCTION

The world is in a state of information overload. Humans are in capable of effectively utilizing the abundance of information currently available. Knowledge information systems aims to efficiently identify, control and leverage information within industries. An effective knowledge culture development programme must motivate employees to learn to manage information and knowledge assets more efficiently. To deliver a sustainable solution the programme must be self-organizing where employees themselves will transmit knowledge and take accountability for the program’s success. Out concept will be based on interpretation of best practices as per the findings of the continued research we are doing this environment.

The concept of a “digital firm” refers to a firm with substantial use of information technology to enhance its ability to sense and respond to its environment. The emergence of information economy and the digital firm in which the major source of wealth and prosperity is the production and Over the years with experience and expertise achieved the conventional factors of production viz., land, labour and capital have become easier to handle. Knowledge has emerged as the fourth and the most important factor. This factor has been in existence for centuries. However it plays a vital role in today’s global economy. It is decisive in that it distinguishes a successful enterprise form a struggling-to-survive enterprise. Though the importance of knowledge is widely acknowledged, there is a lack of comprehensive approach to
managing knowledge for profit and performance maximization.

The knowledge–oriented work can be identified and appreciable at all times at different levels of management. Traditional industrial workers are gradually replaced by knowledge workers, who represent the largest group within corporate organizations (Maier and Sameting 2002). A 1997 Delphi study in the U.S revealed that two-thirds of American companies consider between 60% and 100% of their headcount to be knowledge workers (Delphi 1997). A knowledge driven organization must be a learning organization. Peter Senge is the pioneer of learning organization. As per his theory “it is a place where people continually expand their abilities to create results, where new and innovative patterns of thinking are nurtured and where people are continually learning how to learn together”. Applying knowledge information systems to existing tasks leads to productivity enhancements while using it to develop novel undertakings is referred to as innovation.

The value of a firm’s products and services is based not only on its physical resources but also intangible assets. As knowledge becomes a central productive and strategic asset, organizational success increasingly depends on the firm’s ability to produce, gather, store and disseminate knowledge. Some firms can perform better than others because they have better knowledge about how to create, produce, and deliver products and services. The firm knowledge is difficult to initiate, unique, and can be leveraged into long-terms strategic benefit. The knowledge management systems can collect all relevant knowledge and experience in the firm and make it available wherever and whenever it is needed to support the business process and decision making in the organization. Producing unique products or services at lower cost than competitors is based on knowledge.

The primary steps in decision-making process are

(1) Identifying the business situation,
(2) developing the alternatives solutions,
(3) evaluating alternative solutions in terms of feasibility, satisfactoriness, and potential consequences,
(4) selecting the best alternative,
(5) implementing the chosen alternative,
(6) collect the feedback to refine the organization in terms of efficient and effectiveness.

II. FUNCTIONS OF KNOWLEDGE INFORMATION SYSTEMS IN THE ORGANIZATION

Knowledge information system (KMS) support processes for discovering and codifying knowledge, sharing knowledge, and distributing knowledge, as well as processes for the creating new knowledge and integrating it into the organization.

1. Creating knowledge: Knowledge management systems provide knowledge workers with different management tools like graphics, analytical, communication as well as access to internal and external sources of data to help them to generate new ideas.

2. Discovering and Codifying Knowledge: Organizations are using artificial intelligence technology to capture individual and collective knowledge and to codify and extend their knowledge base.

3. Sharing knowledge: The employees’ access and work simultaneously on the same document from different locations of the organization.

4. Distributing knowledge: Office systems and communication tools can distribute documents and other forms of information among information and knowledge workers and link offices to other business units inside and outside of the organization.

The Knowledge management Best practices are the most successful solutions or problem-solving methods that have been developed by a specific organization or industry. The improvement of work practices, the knowledge can be preserved as organizational memory to train future employees or to help them with decision making. Information systems can also provide knowledge networks for linking people so that individual with special areas of expertise can be easily identified and knowledge can be shared among them. The user can examine the information systems for supporting information and knowledge work.

The Knowledge Management may be viewed from each of the following perspectives.
1. Techno-centric: Focus on technologies, ideally those that enhance knowledge sharing / growth, frequently and technology that does fancy stuff with information.

2. Theoretical: Focus on the underlying concepts of knowledge and truth.

3. People view: Focus on bringing people together and helping them exchange knowledge.

4. Process view: Focus on the processes of knowledge creation, transmission, transformation, and others.

5. Ecological: seeing the interaction of people, identity, knowledge and environmental factors as a complex adaptive system.

III. OBJECTIVES OF KNOWLEDGE INFORMATION SYSTEMS

1. To understand the process involved in capturing and sharing an organization’s assets, both tacit and explicit.

2. To identify and select technology tools for the stages of creation, acquisition, transfer, sharing and use of knowledge.

3. To understand the role of technology in achieving the goals of knowledge information systems.

4. To exploit the techniques of knowledge audit and systems analysis in identifying and characterizing organizational knowledge and information needs.

5. To understand how knowledge information supports business objectives in organizations.

6. To identify Knowledge Information System implementation areas.

IV. TYPICAL KNOWLEDGE RETRIEVAL SYSTEM:

Knowledge represented in structured way is consistent with human thoughts and is easily understandable. The conceptual framework of a typical knowledge retrieval system and the main process can be described as follows.

1. Knowledge discovery: Discovery knowledge from sources by data mining, machine learning, knowledge acquisitions and other methods.

2. Query Formulation: Formulating queries from user needs by user inputs, the inputs can in natural languages and artificial languages.

3. Knowledge Selection: Selecting the range of possible related knowledge based on user query and knowledge discovered from data and information sources.

4. Knowledge Structure Construction: Reasoning according to different views of knowledge domain knowledge, user background etc. In order to form knowledge structures. Domain knowledge can be provided by expert systems. User back ground and performance can be provided by user logs.

5. Exploration and Search: Exploring the knowledge structure to get general awareness and refine the search. Through understanding the relevant knowledge structures users can search into details on what they are interested in get the required knowledge.

6. Knowledge Structure Reorganization: Reorganizing knowledge structures if user need to explore other view of selected knowledge.

7. Query Reformulation: Reformulating the query if the constructed structures cannot satisfy the user needs.

Theories and Technologies Supporting Knowledge Retrieval Systems:

It is generally believed that new ideas repackaging or reinterpretation of old ones. As a new research field, Knowledge retrieval can draw results from the following related theories and technologies.

- Theory of Knowledge: Knowledge acquisition, knowledge organization, knowledge representation, knowledge validation, knowledge management.
- Machine Learning and Knowledge Discovery: Preprocessing, classification, clustering, prediction, post processing, statistical learning theory.
- Psychology: Cognitive psychology, cognitive informatics, concept formation and learning, decision making, human-computer interaction.
- Information Technology: Information theory, information science, information
retrieval, database systems, knowledge-based systems, rule-based systems, expert systems, decision support systems, intelligent agent technology.

- Linguistics: Computational linguistics, natural language understanding, natural language processing.

V. ROLE OF KNOWLEDGE INFORMATION SYSTEMS IN THE ORGANIZATION:

A knowledge information system refers to the set of processes developed in an organization to create, gather, store, transfer, and apply knowledge. Information technology plays an important role in knowledge management by supporting these business processes for creating, identifying, and leveraging knowledge throughout the organization. Once information has been collected and organized in a system, it can be leveraged and reused many times. Knowledge is increasingly becoming the key factor for business and social transformation. The knowledge can be categorized into explicit and tacit knowledge. The explicit knowledge can be expressed in terms of words or numbers that will be stored, disseminated and shared easily. The tacit knowledge cannot be articulated and highly subjective, that will be generated with practical experiences, personnel experiences, feelings or intuitions that are difficult to imitate. Capturing tacit knowledge and externalizing it, that is transforming it into explicit knowledge is always a complex task. Some organizations have created explicit knowledge management programs for protecting and distributing knowledge resources that they have identified and for innovation of new sources of knowledge. Knowledge management increases the ability of the organization to learn from its environment and to incorporate knowledge into its business process.

Knowledge Management (KM) refers to a range of practices and techniques used by organizations to identify, represent and distribute knowledge, know-how, expertise, intellectual capital and other forms of knowledge for leverage, reuse and transfer of knowledge and learning across the organization. Knowledge Management programs are typically claimed to be tied to specific organizational objectives and are intended to lead to the achievement of specific targeted results such as improved performance, competitive advantage, or higher levels of innovation.

Knowledge Management is an evolving discipline. While knowledge transfer has always existed in one form to another, for example through on-the-job discussions with peers, formally apprenticeship, the maintenance of corporate libraries, professional training and mentoring programmers, and–since the late twentieth century–technologically through knowledge bases, expert systems, and other knowledge repositories, KM programs claim to consciously evaluate and manage the process of accumulation, creation and application of knowledge which is also referred to by some as intellectual capital. KM has therefore attempted to bring under one rubric various standards of thought and practice relating to:

- Intellectual capital and the knowledge worker in the knowledge economy
- The idea of learning organization;
- Various enabling organizational practices such as Communities of Practice and corporate Yellow page directories for accessing key personnel and expertise.
- Various enabling technologies such as knowledge bases and expert systems, help desks, corporate intranets and extranets, Content Management and Document Management.

The information work is term of creating the information or processing the information. The information work performed by the information workers. The information workers are classified into two categories: The first category of information workers is Knowledge Workers, who can create the knowledge and information in the organization. Ex: Researchers, designers, architects, decision makers and judges. The second category i.e. Data Workers who can process and disseminate information. Ex: secretaries, bookkeepers and data processors. The Knowledge workers are generally highly educated and experienced than Data workers. The Knowledge workers are the decision makers in their regular routines. The innovation of knowledge programs organized by the special manager called as Chief Knowledge Officer (CKO). The chief knowledge officer is a senior executive who is responsible for the firm’s knowledge management program.
The data-information-knowledge-wisdom hierarchy is used in information sciences to describe different levels of abstraction in human centered information processing. Computer systems can be designed for the management of each of them. Data Retrieval Systems (DRS), such as database management systems, are well suitable for the storage and retrieval of structured data. Information Retrieval Systems (IRS), such as web search engines, are very effective in finding the relevant documents or web pages that contain the information required by a user.

The storage methods of information in the web, literature databases, and digital libraries are documents, information flows, etc. They are not closely related to the ways that human organize knowledge. People need to find, learn, and reorganize retrieved results to extract and construct knowledge embedded in information.

Information Technology plays the key role as indicated by the strong parallelism in growth between supporting technology and the importance of KM over the past five years. Supporting technology is also referred to as Knowledge Information Systems. The Knowledge Information Systems are not only large centralized databases that simply store bulk of data and documents and additionally provide some functions, but consists of networks of interconnected applications, leveraging existing IT infrastructure to align the knowledge relevant aspects. Management of knowledge is strongly supported by information technology, which offers enabling software instruments. Developing procedures and routines-business processes-to optimize the creation, flow, learning, protection, and sharing of knowledge in the firm is now crucial factor of management responsibility.

The organization is using various intelligent computing techniques to capture individual and collective knowledge and codify and extend their knowledge. These systems are collected as intelligent systems. The Artificial Intelligence is the computer-based system that behaves as humans. The organizations will use Artificial Intelligence for capturing and storing knowledge and expertise such systems would be able to learn natural languages, accomplish the physical tasks (robotics), and emulate human expertise and decision making (Expert systems). The Neural networks are designed to imitate the physical thought process of the biological brain. A neural network consists of hardware or software that attempts to emulate the processing patterns of the biological brain. The neurons network to process very large amounts of data efficiently. A neural net has a large number of sensing and processing nodes that continuously interact with each other.

A knowledge repository is a collection of internal and external knowledge in a single location for more efficient management and utilization by the organization. The knowledge collected from many different sources can be documented with different tools in the form of memos, reports, presentations, and articles can be organized into digitized form and kept in the database for easy storage and retrieval. Repositories are used for administration of information by offering intelligent storage services. A data warehouse is collection of information — gathered from many different operational databases — used to create business intelligence that supports business analysis activities and decision making tasks. An enterprise information portal provides a single point of access to the firm’s knowledge resources, and helps the firm coordinate information and people to make decisions and take action. Example some commercial software including SAP’s knowledge warehouse, lotus notes and Oracle applications etc.

The Knowledge Information Systems and artificial intelligence systems can enhance organizational processes in a number ways. The systems can facilitate communication, collaboration, and coordination; bring more analytical power to bear in the development of solutions; or reduce the amount of human intervention in organizational processes.

CONCLUSION

As the basis of value creation increasingly depends on leveraging firms’ intangible assets, Knowledge Information Systems (KMS) are emerging as powerful sources of competitive advantage. However, the general recognition of the importance of such systems seems to be accompanied by a technology induced drive to implement systems with inadequate consideration of the fundamental knowledge problems that the KMS are likely to solve. This paper contributes to the stream of
research on Knowledge Management Systems by proposing an inductively developed framework for this important class of information systems and provides a means to explore issues related to KMS and unifying dimensions underlying different types of KMS. The size and diversity of networks, the maintenance of knowledge flows and the long term effects of the use of KMS provide a window into work in a number of reference disciplines that would enrich the utility of KMS and also open up fruitful areas for future research.

REFERENCES:

ATM Security By Using Fingerprint Recognition

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Abstract- The objective of this paper was to provide a reliable environment to the customer for secure monetary transactions by use of finger print to authenticate the user when larger amount of money is to be withdrawn and then asking the user to enter the password to access the account thus providing twofold security since finger prints cannot be stolen and establish explicit link to the identity to the user. The working of these ATM machine is when customer wants to withdraw a big amount, at that time user figure print is asked. Figure print scanning will be done only limited number of times and if the person is not able to login in that number of times; the system will temporary blocked the account. At the same time a message will be sent to user mobile and is asked for a particular 4-digit code. Only after sending the particular code that the user account will be activated again. This makes the developed ATM software more secure as compared to the software that authenticates the user merely by using a PIN or password.

Keyword- ATM terminal, finger print recognition, PIN.

I. Introduction

Automated Teller machines (ATMs) use continues to grow at a staggering rate. And the customers are authenticated by a password or by entering a personal identification number (PIN). PIN can be acquired and used for unauthorized access thereby making the user account insure for monetary transaction. Security can be increased by the use of biometric authentication methods in which user is checked by its intrinsic physical or behavioral traits like fingerprint etc. If user's bank card is lost and the password is stolen, the criminal will draw all cash in the shortest time, which will bring enormous financial losses to customer. How to carry on the valid identity to the customer becomes the focus in current financial circle [1]. As current ATM systems installed at various locations to be used by bank customers are less secure since the user is using a password or PIN. To authenticate and identify that any unauthorized transactions cannot take place some reliable authentication scheme must be employed by banks. Keeping this thing in mind we develop finger print recognition based ATM software that will asked user’s finger prints when a big amount (let say Rs 15000) is to be withdrawn. So in case if user card is stolen and password or PIN is hacked then only up to that specific amount can be stolen because if the hacker wants to withdraw more money he will be asked for finger print recognition. So this makes the developed ATM software more secure as compared to the software that authenticates the user merely by using a PIN or password. And with the development of biometric solutions for the ATMs there is no need to remember PIN numbers.[2]

II. The Characteristics

As our proposed ATM system is based on finger print recognitions. Matching algorithms are used to compare previously stored templates of fingerprints against user fingerprint for authentication purposes. In order to do this either the original image must be directly compared with the user image or certain features must be compared. The basic functions of this ATM are:

- Figure print verification: If the user want to withdraw a big amount at that time he was asked for the finger print scanning.
- Figure print validation: The system will compare the current finger print with the finger prints of all the users in the databases for a match.
- Warning message: the system will try to scan the user finger print for a limited number of time(let say 5 times) and after that it temporary blocked the account and a message will be displayed on the user mobile that will asked for a particular 4
bit code (that code is given to every customer by the Bank). Only after sending that particular code that the user account will be activated again.

- Two discriminate analysis methods: Both fingerprint and password recognition can be used for the system.

A. **Hardware design**

Required hardware used should be easy to maintain, implement and easily available. Proposed model consists following parts: Finger print scanner; Intel Dual Core CPU, 2.00GHz; Random Access Memory (RAM), 512MB; Hard disk space; Computer (LCD); Keyboard.

Finger print scanner will be used to input fingerprint of customers into the computer software. LCD will help in making the transaction. It will input fingerprint, will process it and extract features for matching. After matching it will update database entries of the customers and keep a record of any transaction made by him/her.

B. **Software design**

Our system integrate biometric identification into normal, traditional authentication technique use by electronic ATM machines nowadays to ensure a strong unbreakable security and non-repudiate transactions. In order to increase the security we are using the combination of authentication of card, PIN and user fingerprint. The proposed software design involves two phases namely: registration phase and verification phase.

1) **Registration phase**- the objective of this process is to create the profile of the user. This process is carried out by the administrator of the system. As shown in fig 1. The process consists of the following two steps:

- **Sample Capture**- the user allows three biometric readings by placing a finger on a fingerprint reader. The result will be combine and compare together and will influence the level of accuracy at the time of validation. all the fingerprint are not stored; the technology will analyzed and measured various data points unique to each individual. The numbers of measured data points varies in accordance to the type of device.

- **Conversion and Encryption**- the individual measurements and data points are converted to a mathematical algorithm and encrypted. These algorithms are complex and cannot be reversed engineered to obtain the original image. This algorithm is further stored in the database or server.

2) **Identification and verification process**- once the individual has been enrolled in a system he/she can start the use of biometric technology to have access to his account via the ATM.
machine to authorize transaction. As shown in Fig:1

- **Identification and Verification** - First the system required all the essential data, like ATM card, pin number etc. but whenever user want to access a large amount of money that limit will be set by the bank, in that case the system required user’s fingerprint. If all the recognition is right, the user can access the ATM Terminal. If Authentication Failure then in that case the system again required user’s fingerprint but only for the limited number of times, this number will also be set by the bank. The user will get limited number of chances to access his /her account. If user still fail to login, the system will block user account and send the alert message to the Account holder and Bank with the new 4 bit code. If user is authenticated he will again get a chance to login. As if he is not an authenticated user he will not be able to enter the 4 digit code. This whole process will only fail if person got some one ATM card, its PIN number and his / her mobile device. This makes the developed ATM software more secure as compared to the software that authenticates the user merely by using a PIN or password. (As shown in figure)

**IV. Results & Conclusions**

Integrating ATM system with the biometric authentication techniques is a solution to avoid the fraud. This proposed system can be used for secured monetary transaction by various organizations. Organization can used this system to allow only selected and trusted users to make financial transaction. Biometric authentication ensures that a person is actually present rather than their cards and password without requiring the user to remember anything. Since this proposed system authenticated its user by fingerprint, it is difficult for an unauthorized user to have access to account that can be used for financial transaction. And in some case if any unauthorized person hacked the PIN or password then he can withdraw only a limited amount of money as whenever he exceed the limit the system will asked for his fingerprint. And if the person unauthorized the system will temporary blocked the account and will sent a warning message to the account holder, and when the account holder sent the specific authorized code (that will be given by bank) only after that the user account will be activate again.

As this proposed system involves additional hardware cost such as scanner but it increase the security of ATM as compared to the currently ATMs. As we know that full secure system does not exit but with this proposed work we can control the fraud to some level because now the hacker can withdraw only a limited amount of money.

**V. References**


Mbed Microcontroller Based Temperature Measurement using 1N4148 Diode sensor

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Abstract—Temperature is one of the most common real-world characteristics that the system needs to measure. Many industrial processes, like steel manufacturing and semiconductor fabrication requires temperature measurement. The temperature measurement can be carried out by using sensors like thermometer, Thermistors, Thermocouples, RTD, Diodes, IC sensors etc. Compared to thermistors, silicon diode gives most efficient temperature measurement, because of their smaller size, accuracy and consistent response to temperature variations. A silicon diode 1N4148 is used as a temperature sensor for this circuitry. This diode has the serious drawback of changing its basic characteristic in the influence of a varying ambient temperature. Fortunately it can be used as a temperature sensor. In the signal conditioning circuit of this paper we will be using quad op-amp, a voltmeter and a few resistors with an Arm controller and a LCD to display Temperature. Using this circuit the temperature measurement can be done between 0°C and 100°C accurately.

I. INTRODUCTION

Thermometer readings are not available in electrical form. Thermistors are nonlinear devices and are not be suitable for wide range of temperatures. And thermocouples are not economical for general usage. RTDs are even more in cost than a diode.

1N4148 diode, like most other semiconductor components, has the serious drawback of changing its basic characteristic in the influence of a varying ambient temperature [1]. But paradoxically, this property in fact has turned into a boon for the construction of various electronic temperature controllers and so this diode can be simply used as a temperature sensor [2].

Compared to thermistors, silicon diodes are most efficient because of their smaller size and accuracy and consistent response to temperature variations. The only drawback with these components is that they are likely to get damaged at temperatures above 150°C [3]. But this maximum range satisfies most of its application requirements.

The diode 1N4148 is comfortably able to produce a linear and an exponential voltage drop across itself in response to a corresponding increase in the ambient temperature [4]. This voltage drop is around 2mV for every degree rise in temperature [5]. This 1N4148 diode has reverse recovery time of 4ns and forward recovery time of 20ns which suits our need to show instantaneous changes in temperature.

The motto of this circuit is to show the linear variation of forward cutoff voltage as a function of temperature to display it as a measure of temperature. Most difficult task of this circuit is to display a decrementing quantity (diode forward voltage) as an incrementing quantity (temperature). To achieve this task an inverting amplifier has to be employed. To display 0°C as 0v there is a need to nullify the diode forward voltage at 0°C hence a summing amplifier with one of input as a dc offset voltage and another input as diode forward voltage has employed.

The circuit of this temperature monitor also incorporates 1N4148 as the sensing device, and is built around a single IC LM324N. Once built and installed, it will provide a reasonably accurate reading of room’s varying atmospheric temperatures, so that now you can remain
informed 24/7 about the minutest temperature changes taking place inside a house.

II. THERMISTOR

A thermistor is a type of resistor whose resistance varies significantly with temperature, which is due to the variation in the number of available charge carriers and their mobility. When the temperature increases, the number of charge carriers increases and the resistance decreases, thus yielding a negative temperature coefficient. This dependency varies with the impurities. The semiconductor achieves metallic properties and shows a positive metallic coefficient over a limited temperature range.

Mathematical modeling of Thermistor:

Steinhart–Hart equation:

\[ \frac{1}{T} = A + B \log R + C(\log R)^3 \]

Where \( T \) = Temperature; °k,
\( R \) = Resistance of thermistor;
\( A, B, C \) = Curve fitting constants

The equation is often used to derive a precise temperature of a thermistor since it provides a closer approximation to actual temperature than simpler equations, and is useful over the entire working temperature range of the sensor. Steinhart–Hart coefficients are usually published by Thermistor manufacturers.

Steinhart–Hart coefficients can also be derived. Three accurate resistance measurements are made at precise temperatures, and then the coefficients can be derived by solving three simultaneous equations.

Signal conditioning circuit for Thermistor:

Linearization: It is possible to use a Thermistor over a wide range of temperature, only when the system designer can tolerate a lower sensitivity to achieve improved linearity. One approach to linearizing a Thermistor is simply shunting it with a fixed resistor. Paralleling the Thermistor with a fixed resistor increases the linearity significantly. As shown in figure

![Fig.1 Linearization of Thermistor](image)

III.1N4148 Diode Sensor

Fig.3 block diagram

Block Diagram Description:
Constant Voltage:
Here a constant voltage source is employed for generating a offset voltage which is required to eliminate the offset voltage incurred in the summing amplifier.

Offset generation:
The exact offset voltage is generated by using a potentiometer in parallel to the zener diode and taping it.

Buffer Amplifier:
Two buffer amplifiers are used in the circuitry in order to provide constant voltage from the source irrespective of their impedances.

Powering Sensing element:
An inverting amplifier is used as powering element for the sensor 1N4148 in order to make the decrement in forward cutoff voltage as an incrementing quantity for processing.

Summing Amplifier:
A summing amplifier in inverting configuration is used to eliminate the dc offset present at the output of inverting amplifier so that the 0°C can be shown as 0v output. Here we arrive a condition that a direct multiplication of output voltage by a factor of 100 will directly yield the temperature.

Circuit Diagram:

Simulation result:

Fig. 4 Signal conditioning circuit

Simulation result:

Applying KVL at node 3.
\[
\frac{7.5-v_1}{R_1} + \frac{v_0-v_2}{R_2} = 0
\]
\[
\frac{7.5}{R_1} + \frac{v_0}{R_2} = \frac{1}{R_1+R_2} \quad \text{.... (1)}
\]

Applying KVL at node 2.
\[
\frac{v_2 - 0}{R_3} + \frac{v_2 - v_{01}}{R_4} + I_d = 0
\]
\[ \frac{v_2 - 0}{R_3 + R_4} - \frac{v_3}{R_4} + I_d = 0 \quad \text{.... (2)} \]

Id=600mA (practical observation)

By virtual ground

V1=V2=592.0349-2.22T

=600-2.22T (practical observation)

\[ \frac{7.5}{R_1} + \frac{v_0}{R_2} = \frac{600 - 2.2 \times T}{R_1 + R_2} \]

Assuming \( R_1 = R_2 \)

\[ \frac{7.5}{R_1} + \frac{v_0}{R_1} = \frac{600 - 2.2 \times T}{2 \times R_1} \]

7.5+V_0= (600mV-2.2TmV)*2

V_0= 1.2-7.5-0.0044T

**Summing amplifier:**

![Summing Amplifier Diagram]

\[ V_{d} = 0.6 - 0.00225 \]

\[ V_{out} = \frac{R_5}{R_6} \]

\[ V_{out} = \frac{E(V_1) + E(V_2)}{2} \]

Assume R7=R5

For the output voltage to be varying from 0 to 1v

\[ \frac{R_2}{R_6} = 4.5454 \]

Assuming R7 to be 150K from available resistor in market

150k*0.22=R6

R6=33K

R7=R5=150K, R6=33K

**Signal conditioning Circuit description:**

We have chosen circuit such that an output of 1v corresponds to a Temperature of 100°C and an output voltage 0v corresponds to 0°C. This allows temperature to be measured to a precision of 0.2°C

Here signal conditioning circuit is built around a low power quad operational amplifier (LM324N) and it is used for achieving better accuracy [6]. The quad op-amp is powered from a dual rail supply in the range +/- 9v to +/- 15v

The zener diode is used to provide a constant voltage of 7.5v from which a voltage of -3.0765v can be ‘tapped off’ (practically observed for the given circuit arrangement) using potentiometer. This voltage is buffered by ‘op-amp A’ before presenting it to ‘op-am-D’, which is an inverting voltage summer. The full voltage across the zener diode is buffered by ‘op-amp B’ before presenting it to ‘op-amp C’. The current supplied to IN4148 diode by this circuit is equal to \(-V1/12\) (V1 is here -7.5v)

Resistors around op-amp C are carefully matched to assure constant current. The voltage across the IN4148 diode is fed to op-amp D for
implementing an analog voltage operation and the relationship is given by the following equation.

\[ V_{\text{out}} = \frac{R_4}{R_3} V_{\text{offset}} \]

Here \( V_{\text{offset}} = -3.076\text{v} \) (nearly), \( R_3 = R_4 = 150\text{K} \), \( R_4 = 33\text{K} \) (nearly).

**Arm controller:**

Arm controller [7] has a 6 input multiplexed, 12-bit DAC. One of its 6 channels has been used for temperature measurement and we use the others for future need within the same project. The ADC has a resolution of 0.8056 mV. In earlier times we needed to interface a separate ADC to microcontroller but in the case of Arm controller task of interfacing has been reduced [8].

The mbed NXP LPC11U24 Microcontroller in particular is designed for prototyping low cost USB devices, battery powered applications and 32-bit ARM® Cortex™-M0 based designs. It is packaged as a small DIP form-factor for prototyping with through-hole PCBs, strip board and breadboard, and includes a built-in USB FLASH programmer. Unlike the conventional Microcontrollers used for temperature and other quantity measurement Arm controller implementation is very easy [9]. The arm controller can be used to send temperature readings to a pc through a wired connection which facility is available in some other systems like SHT11/71 INTELLIGENT SENSOR [10].

**NXP LPC11U24 MCU FEATURES:**

- Low power ARM® Cortex™-M0 Core
- 48MHz, 8KB RAM, 32KB FLASH
- USB Device, 2xSPI, I2C, UART, 6xADC, GPIO
- Prototyping form-factor

40-pin 0.1” pitch DIP package, 54x26mm
5V USB, 4.5-9V supply or 2.4-3.3V battery

The mbed Microcontrollers are supported by the mbed.org developer website, which includes a lightweight Online Compiler.

**Calibration of circuit:**

The circuit has been of good ease to implement as it contains only one potentiometer to be adjusted to get the desired operation. We need to adjust the potentiometer at a single temperature to get output as Temperature (in °C) divided by 100 as our scaling factor is 100.

**IV.EXPERIMENTAL SETUP**

The above described signal conditioning circuit has been implemented practically and the setup is as shown below. The LCD and sensor are included in a closed snap shot.
Sensor:

\[
\text{Fig. 10 Sensor}
\]

V. RESULT ANALYSIS

<table>
<thead>
<tr>
<th>Temp in °C</th>
<th>Calibrated output in Volts</th>
<th>Practical temp output= (V_{\text{out}}\times 10)</th>
<th>Error in %</th>
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<td>0.895</td>
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<td>0</td>
</tr>
</tbody>
</table>

\[
\text{Fig. 11 Variation of percentage of error for Thermistor and 1N4148.}
\]

VI. CONCLUSION

When a thermistor is used as a temperature sensing element the obtained percentage of error is in between 1% and 9.5% for the range of 0° c and 60° c. By using a diode 1N4148 as sensing element we were able to measure the room temperature accurately with nearly 2% error at any temperature between 0° c and 100° c. This diode is a high speed switching device so that it is possible to show the instantaneous changes accurately. In this paper we are using an Arm controller which is programmed for displaying temperature as there is no provision of providing a gain of 100 using an analog circuit to display 1V as 100° c. So the calibration is performed using an Arm microcontroller. And it is also easy for beginners to start with such easily programmable microcontroller. And it also has good resolution to display temperature accurately.
REFERENCES
[9] “Humidity and temperature measurement system using a low-cost Universal Transducer Interface”.

Appendix
/* programme to display temperature on lcd
using arm controller*/

#include "mbed.h"

#include "TextLCD.h"

TextLCDlcd(p19,p20,p21,p22,p23,p24);//rs,e,d0 ,d1,d2,d3

AnalogInAin(p18);
float temperature;

int main()
{
  while(1)
  {
    temperature=Ain*3.3*100;
    lcd.printf("TEMPERATURE IS \n %1.2f C",temperature);
    wait(1);
    lcd.cls();
  }
}
Data Transfer
Secure Remote Network Monitoring System

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School Of Computer Science, Lingaya’s University Faridabad (India)

Abstract-In this paper we have proposed Remote Monitoring is designed and developed with the end user in mind, user in a similar way. The features included in the Remote Monitoring package are a jump further on of other similar fields. A remote monitoring, also called a LAN Controller, is a desktop application designed for use on a small office environment. A Remote Monitoring is optimized so as to display client pc which is connected in local area network. Remote Monitoring software must be small and efficient to accommodate the low memory capacity. We trying offer the expertise and ability to control client pc in a small office environment, easily send file server to client and also send warning message and multiple line messages to client. This system is useful for those offices which have multiple users in different locations and admin want to monitor all users in an office. This Application also provides server monitor all clients in an office. In this, server is able to send file to client and also send warning message and multiple line messages to client and server is also able to shut down client pc. This facility is provided in which place where local area network is available. The system will mainly be developed by targeting the information seek peoples such as professional, business person etc. to fulfill their requirements.

Keywords—Remote network monitoring system, LAN, RMON, SMON, IETF

I. INTRODUCTION

Research on remote network monitoring systems was developed by the IETF to support monitoring and protocol analysis of LANs. The first Original version about Remote monitoring system referred as RMON1 can focused on OSI Layer1 and OSI Layer2 information in Ethernet and Token ring networks. It can be extended by RMON2 which add support network and application layer monitoring and by SMON which adds support for switched networks. It is an industry standard specification that provides much of the functionality offered by proprietary network analyzers. In RMON agents are built into many high end switches and routers. But in our work the expertise and ability to control client pc in a small office environment, easily. Send file server to client and also send warning message and multiple line message to client.

This paper contains four sections. The first section describes problems designers meet when creating a Remote network monitoring system. The second section describes various approaches taken in designing a Remote network monitoring system; it also describes some real-life systems. The third section tries to describe the future. The fourth section concludes this paper.

II. PREVIOUS RELATED WORK

While email is now widely accepted, it only represents a developing stage that will over the next five years develop into entirely new forms of communications that more directly and efficiently suit the needs of businesses and organizations. Existing system has the ability to split attachments into chunks. This can be both annoying, if it was setup without knowing, or useful. Some Service Providers have a maximum limit on the size of emails. So your email Service provider may only allow a maximum of 1mb and will reject everything above that. Have you noticed when you try and click on a hyperlink in an email, nothing happens. This could be a number of reasons. Virus Software is preventing links from opening. Default Browser is not set correctly. Windows File Association for web pages could be wrong In an existing system I have found that as many times user receive an email from an address and user just delete it. It may be as simple as just unsubscribing from the mailing list. However, sometimes you may just want to block the email
address so it doesn’t end up in your Inbox. Most users do not want that their email and password should be saved in a particular site. It has a large security problem. In the proposed system developer will use the RSA encryption method; here messages are translated into sequences of integers. This can be done by translating each letter into an integer, as is done with the Caesar cipher. These integers are grouped together to form larger integers, each representing a block of letters. Other existing system has trouble downloading new mail. These problems can occur if you do not compact the Inbox. The mailboxes require intervallic protection to avoid such problems from occurring. By the time the problems occur, compacting the Inbox, though still required, may not be sufficient to resolve the problem. In the proposed system this kind of problem won’t be happen. Because the developer will use the actual methodology of SMTP and POP3 protocol. Same mail messages continue to be downloaded from POP3 account. If your POP3 mail account is configured to “leave a copy of messages on the server” it sometimes happens that OE downloads the same messages each time you send and receive. This is caused by a damaged Pop3 file in your Identity’s store folder. I had found sometimes that many existing system has trouble to downloading new mail. When this happens, you may receive new messages that are empty when you try to open them or multiple entries for the same new messages in the inbox. One common problem in some existing system is that users experience is an inability to send messages. If you are unable to send messages, try deleting the outgoing mail (SMTP) password for your account.

2. Different Approaches

Research is of prime importance for any successful project development. It gives movement to developers’ ideas and provides for the theoretical background for any project development. This section will with the developer’s existing knowledge and secondary research. The section will also present a domain analysis any such existing systems.

Email is the most preferred communication medium for businesses around the world today and providing protected and pervasive access to it is the key to increased productivity of any successful business. Smart Mail provides intelligent access to Email, contacts directory and calendar. Interaction with Smart Mail is multi-modal providing hassle-free access to the user. Smart Mail alerts the users based on their alert rules. Users can access Smart Mail through any mode/device of their choice to read, reply, compose, and forward emails and attachments.

E-mail is the most widely used application and it has become the application that users spend much of their time using it \([1, 2]\). E-mail clients are used every day in our lives and they can be the reason of buying personal computers. Therefore, Duchenaut and Bellotti called it habitat \([3]\). E-mail is designed to enable none-face to face communication but because it is widely used nowadays it is being used for additional functions which e-mail is not designed for, therefore it is called overloaded \([4]\). Many studies showed that the number of e-mail messages rapidly grows. For example, the University of California stated that about 31 billion e-mail messages have been sent in 2007 and this number might be doubled in 2010. In other study, it has been shown that the average user gets around 49 e-mail messages a day while high volume users can get more than hundred \([1]\). It has been shown that the average user gets around 49 e-mail messages a ‘day. While high volume users can get more than hundred. By leaving these e-mail messages in the e-mail inboxes they will be very difficult to use and they will be cluttered.

Although e-mail applications are widely used and many studies have been performed but it has not been significantly treated and researched in term of usability aspects. Although most of the e-mail clients use the textual table view to presents the e-mail messages, different problems are exist when browsing large
amount of e-mail messages. For example users will take long time to locate a specific e-mail message.

Filtering and using folders has been proposed and suggested to be the way of organizing the e-mail inboxes which will make the navigation easier. Many studies showed that using folders and filtering features has many problems. Auto classification and automatic folders creation has been proposed to be a solution, but it has proven error prone [1]. Apart of using filtering and auto classification of e-mail messages many studies have been performed in order to provide an alternative approach to the textual table approach. Most of these studies used different visualization techniques with the information hiding concept in order to present the e-mail messages and data.

Research is a significant attribute in determining the accomplishment of a project, as it provides complete coverage of knowledge, better insights and understanding of the required area. The type of research to be applied in this project will be mainly primary researches, such as questionnaires. Specific interviews with some of the experts in the mail system will be essential in order to gather the correct and accurate information required. As Smart mail is a mail client system is still advanced system, there is not much information from the television or books about it, and the only solution is to gather information from the student who have been active to using the mail for some time.

The development of this project requires a deep insight in many new concept of software development. Major research area for the development of the project can be:

- Key concepts of client server architecture.
- Concept of software design methodology.
- Usability principles.
- The advanced concept of database management.
- Understanding the very concept of Email client.
- To know and implement the protocols for mail client development.
- To have a huge ground exposure to large databases.
- To understand the concept of design issues and user feedback.
- To understand the concept of SMTP and POP3 protocol.
- To understand the concept of RSA Encryption method.

Preliminary researches are being carried with the help of the following sources. As research work progress more items will be added to the list. The proposed system is being developed for mail system. So, a study of the various principles of this field is required.

As the email is received by the intended recipient, the email is unencrypted as it passes through the recipient’s secure-mail hub or when it is opened if the recipient is using the software version, Secure-mail. By reviewing the previous research and studies that have been performed in browsing e-mail data it has been inferred that using graphics alone is not sufficient for enhancing e-mail usability. Information hiding is not a good solution for reducing the graphical complexity because it might hide critical information. This information can be presented using other modality such as the auditory channel. The combination of different modalities can enhance the usability of any information system. The e-mail message based on the basic e-mail properties and presents the other using different modalities. The anticipated benefits of this tool are expected to be significant and effective because the e-mail applications are widely used.

3. Similar Existing Systems:
There are mail client services that are available in the market today. Given below is a brief description of some of the most popular mail clients.

**Hardware requirement:** The hardware needed for this project are listed below. The specification of the PC is:

<table>
<thead>
<tr>
<th>Processor</th>
<th>Intel Pentium 4-1.3GHz or faster or AMD</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ram</td>
<td>512 MB DDR-Win XP</td>
</tr>
</tbody>
</table>
Software requirement: In order to carry out the development of this project, many software are needed, these include Microsoft Windows XP, which shall be the operating system for the project to be developed; internet connection will also be needed in order to come up with the interactive user interface and objects. List of software

<table>
<thead>
<tr>
<th>Operating system</th>
<th>Windows XP</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Windows vista</td>
</tr>
<tr>
<td></td>
<td>Windows 7</td>
</tr>
<tr>
<td>Licensing</td>
<td>Internet connections required for all computers</td>
</tr>
<tr>
<td>Presentation</td>
<td>Microsoft PowerPoint</td>
</tr>
<tr>
<td>Gantt Chart</td>
<td>Microsoft project 2003</td>
</tr>
<tr>
<td>Diagram</td>
<td>Microsoft Visio 2003/07</td>
</tr>
<tr>
<td>Design</td>
<td>Adobe Dreamweaver 8.0</td>
</tr>
<tr>
<td></td>
<td>Adobe Photoshop</td>
</tr>
</tbody>
</table>

The main tool and programming language tool be used the most for the project are java swing.

III. CONCLUSIONS

In this paper the era of programming by writing code in a code editor is long gone. These days’ programmers are approaching towards rapid code development. Shorter life cycle development and easier maintenance of developed systems can be achieved through different approaches ranging from computer-aided development to rapid prototyping and model development.

The chosen software tool may shorten the production period but, at worst, may considerably prolong it. If the modeling environment is too complex, the programmer will require training and practice time before becoming creative - time that we simply do not have.

The learning experience has been outstanding. It could be as my best project development experience thus far away. It may appear awkward but students do appear not to take their FYP seriously from its start however it doesn’t lose its weight. I am the most interested person when it becomes to marketing a product. This was one of the reasons I chose to make such a system for users, because it’s unique functionality.

This project gave me an outlook to develop a system that will help everyone to creating such a unique system. Yes, of course, my awareness can be cherished by everybody even if he/she is new to computer world. I take this as one of my personal achievements. Honestly, I never perceived the final product in this form as it is at the initiation of the project. Now that I have developed it, it’s hard for me to believe it.

It was the first time I was performing such a passionate research work interviewing people from different background helped me learn a lot. It was the first time I have conducted the questionnaire, and the experience was amazing.
To conclude, I feel very pleased at this moment when my project is done and thankful to all the support I got from everyone.

IV. FUTURE WORK

This paper is based on our ongoing research of Remote network monitoring systems, there is some more groups spread around the world working on the issue today.

Acknowledgment

The authors would like to thank Honorable administrator School of Computer Sciences Dr. Tapas Kumar and respected Faculty members Dr. Anubhav Kumar, Mr. Kiran Kumar & Mr Praveen Gupta of Lingaya’s University, Faridabad, Haryana, INDIA, for providing us the entire necessary infrastructure for higher technical education and valuable support for research & development in the college. Without the facilities at the college, this work would have not been possible.

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Other White Papaers

Design and Development of Motorized Linear Translation Stage using Microcontroller

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Abstract—This paper represents the implementation of motorized linear translation stage using microcontroller PIC16F887 which is used to precisely position an object along a single axis. It includes a moving platform and stationary base joined by a bearing system. Position is controlled electronically with the use of a motion controller. Motorized stages are preferred for high load/high speed applications, as well as applications where manual adjustment may be difficult. To program the microcontroller we first write the program in C-language, and then we convert the code in the hexadecimal code and burn the code within the chip and configure the microcontroller according to our requirement and perform all operations.

Keywords—PIC16 microcontroller, Unipolar hybrid stepper actuator, Character LCD, SPST Switches, Motor driver

I. INTRODUCTION

Motorized Linear Translation Stage is a mechatronics product. In this we are using microcontroller for motion control of unipolar hybrid stepper Actuator which is driven using motor driver circuit. A linear actuator is an actuator that creates motion in a straight line, in contrast to the circular motion of a conventional electric motor. Linear actuators are used in machine tools and industrial machinery, in computer peripherals such as disk drives and printers, in valves and dampers, and in many other places where linear motion is required. Many other mechanisms are used to generate linear motion from a rotating motor.

Typically, an electric motor is mechanically connected to rotate a lead screw. A lead screw has a continuous helical thread machined on its circumference running along the length (similar to the thread on a bolt). Threaded onto the lead screw is a lead nut or ball nut with corresponding helical threads. The nut is prevented from rotating with the lead screw (typically the nut interlocks with a non-rotating part of the actuator body). Therefore, when the lead screw is rotated, the nut will be driven along the threads. The direction of motion of the nut depends on the direction of rotation of the lead screw. By connecting linkages to the nut, the motion can be converted to usable linear displacement. Most current actuators are built for high speed, high force, or a compromise between the two. When considering an actuator for a particular application, the most important specifications are typically travel, speed, force, accuracy, and lifetime.

There are many types of motors that can be used in a linear actuator system. These include dc brush, dc brushless, stepper, or in some cases, even induction motors. It all depends on the application requirements and the loads the actuator is designed to move. For electromechanical linear actuators used in laboratory instrumentation robotics, optical and laser equipment, or X-Y tables, fine resolution in the micron range and high accuracy may require the use of a fractional horsepower stepper motor linear actuator with a fine pitch lead screw.

In our project two tactile SPST buttons are used to position any object in two opposite directions along single axis. Character LCD is to display the distance travelled by the detector which is the alternative of Optical Encoder. The circuit is designed in simulation software Proteus ISIS and the circuit board using Proteus ARES through the process of Layout Designing, Routing, and Chemical Etching etc. MPALAB IDE is used for programming, debugging and loading the hex file in the microcontroller.
II. THEORY

A stepper motor is a brushless, synchronous electric motor that converts electrical pulses into mechanical movement. Every revolution of the stepper motor is divided into a discrete number of steps, and the motor must be sent a separate pulse for each step. The stepper motor can only take one step at a time and each step is the same size. Since each pulse causes the motor to rotate a precise angle, the motor’s position can be controlled without any feedback mechanism. As the electrical pulses increase in frequency, the step movement changes into continuous rotation, with the speed of rotation directly proportional to the frequency of the pulses. Step motors are used every day in both industrial and commercial applications because of their low cost, high reliability, high torque at low speeds and a simple, rugged construction that operates in almost any environment. In the project Unipolar Stepper Motor is used because it’s easy availability and simplicity in the driver circuit.

The unipolar stepper motor has five or six wires and four coils (actually two coils divided by center connections on each coil).

Fig. 1 Unipolar Stepper Motor Windings

The center connections of the coils are tied together and used as the power connection. They are called unipolar steppers because power always comes in on this one pole.

III. STEPPER MOTOR INTERFACING WITH MICROCONTROLLERS

Stepper motors can be used in various areas of microcontroller projects such as making robots, robotic arm, automatic door lock system etc. Here, we will discuss different controlling types (Half step and Full step), Interfacing Techniques (using L293D or ULN2003) to control stepper motor.

A. Step Sequence

Stepper motors can be driven in two different patterns or sequences, namely,

- Full Step Sequence
- Half Step Sequence

1) Full Step Sequence: In the full step sequence, two coils are energized at the same time and motor shaft rotates. The order in which coils has to be energized is given in the table below.

<table>
<thead>
<tr>
<th>Step</th>
<th>A</th>
<th>B</th>
<th>C</th>
<th>D</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
<td>0</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>2</td>
<td>1</td>
<td>1</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>3</td>
<td>0</td>
<td>1</td>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>4</td>
<td>0</td>
<td>0</td>
<td>1</td>
<td>1</td>
</tr>
</tbody>
</table>

2) Half Step Sequence: In Half mode step sequence, motor step angular reduces to half the angle in full mode. So the angular resolution is also increased i.e., it becomes double the angular resolution in full mode. Also in half mode sequence the number of steps gets doubled as that of full mode. Table below shows the pattern of energizing the coils.

<table>
<thead>
<tr>
<th>Step</th>
<th>A</th>
<th>B</th>
<th>C</th>
<th>D</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1</td>
<td>0</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>2</td>
<td>1</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>3</td>
<td>1</td>
<td>1</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>4</td>
<td>0</td>
<td>1</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>5</td>
<td>0</td>
<td>1</td>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>6</td>
<td>0</td>
<td>0</td>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>7</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>8</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
</tbody>
</table>

B. Step Angle

Step angle of the stepper motor is defined as the angle traversed by the motor in one step. To calculate step angle, simply divide 360 by number of steps a motor takes to complete one revolution. As we have seen that in half mode,
the number of steps taken by the motor to complete one revolution gets doubled, so step angle reduces to half.

As in above examples, Stepper Motor rotating in full mode takes 4 steps to complete a revolution. So step angle can be calculated as...Step Angle $\Theta = \frac{360°}{4} = 90°$ and in case of half mode step angle gets half so $45°$. So this way we can calculate step angle for any stepper motor. Knowing stepper motor’s step angle helps you calibrate the rotation of motor also to helps you move the motor to correct angular position.

Movement along a single plane-vertical, horizontal or even rotational is called **translation**. There are several ways to move a stage or platform in this manner. I made single axis horizontal translation stage using PIC16F887 microcontroller and External Linear Unipolar Stepper Motor.

**IV. STEPPER ACTUATORS TYPES**

There are three basic types of stepper actuators. The stepper actuator motor types vary by their construction and in how they function. Each type of stepper actuator offers a solution to an application, in different ways. The three basic types of stepper actuators include the Variable Reluctance, Permanent Magnet and Hybrid Actuators.

A. **Variable Reluctance (VR) Stepper Actuators**

VR stepper actuators are known for having soft iron multiple rotors and a wound stator. The VR stepper actuators generally operate in step angles from 5 to 15 degrees, at relatively high step rates. They also possess no detent torque. When phase A is energized, four rotor teeth line up with the four stator teeth of phase A by magnetic attraction. The next step is taken when A is turned off and phase B is energized, rotating the rotor clockwise 15 degrees; Continuing the sequence, C is turned on next and then A again. Counterclockwise rotation is achieved when the phase order is reversed.

B. **Permanent Magnet (PM) Stepper Actuators**

Permanent Magnet Stepper Actuators differ from variable reluctance stepper actuators by having permanent magnet rotors with no teeth. These rotors are magnetized perpendicular to the axis. When the four phases are energized in sequence, the rotor rotates as it is attracted to the magnetic poles. The motor shown in Figure 6 will take 90 degree steps as the windings are energized in sequence ABCD. Permanent magnet stepper actuators generally has step angles of 45 to 90 degrees and tends to step at relatively low rates, but produce high torque and excellent damping characteristics.

C. **Hybrid Stepper Actuators**

Hybrid Stepper Actuators combine qualities from the both permanent magnet and variable reluctance stepper actuator motors. The Hybrid stepper actuator motors have some of the desirable features of each. These stepper actuator motors have a high detent torque, excellent holding and dynamic torque, and they can operate in high step speeds. Step angles of 0.9 to 5.0 degrees are normally seen in hybrid stepper actuator motors. Bipolar windings are generally supplied to these stepper actuator motors, so a single power supply can be used to power the stepper actuator motors. The rotor will rotate in increments of 1.8 degrees, if the phases are energized one at a time in the order they are indicated at. These Actuator Motors can be driven in two phases at a time to yield more torque. Hybrid stepper actuator...
motors can also be driven by one, then two, then one phase to produce half-steps of 0.9 degree increments. Typically hybrid stepper actuators are more costly due to the combined characteristics.

Fig. 3 Actual View of Motorized Linear Translation Stage

V. SOFTWARE IMPLEMENTATION

Schematic: Proteus ISIS:

Proteus is virtual simulation software. ISIS (Intelligent Schematic Input System) is a part of PROTEUS used to draw circuits and simulate them. There are different modes in ISIS like Component Mode (for making circuit), Terminal Mode (for Vcc and Gnd), Generator Mode, Virtual Instruments Mode, etc. The code is compiled in MPLAB IDE and downloaded in the controller in PROTEUS. Then the circuit is simulated in the ISIS window.
The code is written in MPLAB IDE v8.70

Microchip provides a freeware IDE package called MPLAB, which includes an assembler, linker, software simulator and debugger. MPLAB IDE facilitates programming in assembly as well as in C language.

VI. HARDWARE DESIGN

A. Microchip’s PIC 16F887 Microcontroller

Hardware architecture: Harvard architecture (separate program and data memory). Software architecture: RISC architecture (simpler instructions), 13-bit Program Counter, 14-bit instruction, Only 35 instructions (low-end PIC), 8KB of Program Memory (Flash ROM), 368 Bytes of Data Memory (SRAM), 256 Bytes of Permanent Data Memory (EEPROM), 8-level Stack, In-built 10-bit ADC, In-built USART for serial
communication, Total interrupts: 15, In-circuit serial programming possible, Capture, Compare and PWM Modules, Built-in oscillators with selectable speeds, Flash ROM is re-writable up to 10 lakh times, Once the code is downloaded in the controller, it remains there for at least 20 years

B. ULN2804- Darlington Transistor Arrays

Eight Darlington’s with common emitters, Output current to 500 ma, Output voltage to 50V, Integral suppression diodes, Output can be paralleled, Inputs pinned opposite outputs to simplify board layout

![Fig. 6 Pin Diagram of ULN2804](image)

In electronics, the Darlington transistor (often called a Darlington pair) is a compound structure consisting of two bipolar transistors (either integrated or separated devices) connected in such a way that the current amplified by the first transistor is amplified further by the second one. This configuration gives a much higher common/emitter current gain than each transistor taken separately and, in the case of integrated devices, can take less space than two individual transistors because they can use a shared collector. Integrated Darlington pairs come packaged singly in transistor-like packages or as an array of devices (usually eight) in an integrated circuit.

C. 16 * 2 Character LCD

LCD (Liquid Crystal Display) screen is an electronic display module and find a wide range of applications. A 16x2 LCD display is very basic module and is very commonly used in various devices and circuits. These modules are preferred over seven segments and other multi segment LEDs. The reasons being: LCDs are economical; easily programmable; have no limitation of displaying special & even custom characters (unlike in seven segments), animations and graphics and so on.

A 16x2 LCD means it can display 16 characters per line and there are 2 such lines. In this LCD each character is displayed in 5x7 pixel matrix. This LCD has two registers, namely, Command and Data. The command register stores the command instructions given to the LCD. A command is an instruction given to LCD to do a predefined task like initializing it, clearing its screen, setting the cursor position, controlling display etc. The data register stores the data to be displayed on the LCD. The data is the ASCII value of the character to be displayed on the LCD.

![Fig. 7 Pin Diagram of 16 * 2 LCD](image)

<table>
<thead>
<tr>
<th>Pin No</th>
<th>Function</th>
<th>Name</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Ground (0V)</td>
<td>Ground</td>
</tr>
<tr>
<td>2</td>
<td>Supply voltage; 5V (4.7V – 5.3V)</td>
<td>Vcc</td>
</tr>
</tbody>
</table>
3. Contrast adjustment; through a variable resistor

4. Selects command register when low; and data register when high

5. Low to write to the register; High to read from the register

6. Sends data to data pins when a high to low pulse is given

7. Register Select

8. Read/write

9. Sends data to data pins when a high to low pulse is given

10. Enable

11. 8-bit data pins

12. DB0

13. DB1

14. DB2

15. DB3

16. DB4

17. DB5

18. DB6

19. DB7

20. Backlight VCC (5V)

21. Led+

22. Backlight Ground (0V)

23. Led-

Fig. 8 Screenshot of Proteus ARES Software

D. **PCB Designing**

A **printed circuit board** has pre-designed copper tracks on a conducting sheet. The pre-defined tracks reduce the wiring thereby reducing the faults arising due to lose connections. One needs to simply place the components on the PCB and solder them and last applying of solder flux.
VII. CONCLUSIONS

Motorized Translation Stage is expected to be increasingly prevalent in motion control applications. It is an integrated system that includes concepts of Mechanical and Electronics. Generally, In Motorized Stages Optical Encoders are used to sense the position of object. But, In our project we do some extra part in the programming of microcontroller actually we counted the number of steps required to cover the whole path travel by the motor and according to that we fixed the limits in the steps running by motor in the source code. The EEPROM is used to save the number of steps travelled by the motor permanently so that every time we restart the system it able to recall its position and LCD Display is also used to display the position of object.

VIII. FUTURE WORK

In this project we have worked on half stepping drive of hybrid stepper actuator but for better resolution i.e., minimum incremental motion we can use micro stepping drive of stepper actuator. Using micro stepping we can do micro positioning of any object. In micro stepping phase currents are controlled by fractional amounts, rather than just ON/OFF, resulting in more magnetic equilibrium positions between the poles. In effect, discretized sine waves are applied to the phases instead of square waves.

ACKNOWLEDGMENT

We would like to thank Dr. Ramesh Chand Sharma; Scientist-‘F’, Head (Remote Sensing Technology Division), LASTEC, DRDO, Delhi for providing the opportunity to carry on the research work and successfully implement the concept.

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Abstract— The paper describes basic features and functional implementation of an operational architecture for IPTV content generation and distribution. The architecture is designed to make possible extensive tests. A system overview is also included with its all major operating components. It describes the reasons why customers should choose IPTV for broadcast and video services with different technologies and platforms used for user-friendly interface screens.

Keywords — IPTV, network architecture and communication, Architecture of IPTV, Supporting services and applications of IPTV

INTRODUCTION

Internet Protocol Television (IPTV) is a fast growing technology for the delivery of broadcast streams and other video content services over a safe, uninterrupted and provider managed broadband IP data network. IPTV mainly provides a rich functionality that ranges from the creation, encryption and decryption, parental control and management of video content, to the delivery of digital streams, videos on demand, viewing of stored content, personalized programming guides, and a host of interactive and multimedia services[1][2][12].One of the most attractive features of IPTV is user interface. This approach is unimaginable in a traditional TV broadcasting. End users can request a concrete TV channel or a Video on demand, they have a pay per view offer in EPG (Electronic Program Guide), they can pay for demanded programmed if they want to watch it through the system, there is subscription and rent possibility for advertised items, betting etc [10].

IPTV is distinctly different from “Internet Video” that simply allows users to watch videos, like movie previews and web-cams, over the Internet IPTV technology, uses Internet Protocols (http, rtmp, igmp) integrated with the high speed digital subscriber line (DSL) access technologies (ADSL2, ADSL2+ and VDSL), unleashing exciting new revenue opportunities to the telecom service providers, enabling them to compete effectively in the “triple play” market space with the delivery of voice, data and new services such as broadcast and video on demand (VoD) to residential and business customers in a “best effort” fashion with no end-to-end service management and quality of service considerations.

I. THE MAJOR OPERATIONAL COMPONENTS OF THE IPTV ARCHITECTURE ARE [5]

![IPTV architecture](image)

(1) Content Provider - ‘Content Provider’ can range from live TV networks to niche on-demand content from micro-producers. IPTV systems can deliver an unlimited number of broadcast streams as each television set only require one TV channel connection which can be linked to any other TV source the IPTV operator can provide.

(2) Access Nodes - The ‘Access Nodes’ represents a functionality that receives video streams in various formats, then reformats and encapsulates them for transmission with appropriate Quality of Service (QoS) indications to the wide-area network for delivery to customers.

Broadcast information coming from an antenna or a satellite dish at the national headend is mainly distributed using MPEG-
2 multiprogram transport stream (or MPTS) to the video service node.

(3) **Access Provider** – IP distribution systems transfer media programs from the content sources to viewing devices using various methods such as unicast, multicast and broadcast necessary for the reliable and timely distribution of IPTV data streams from the Service Nodes to the Customer Premises. The Access Provider includes the optical distribution backbone network and the various Digital Subscriber Line Access Multiplexers (DSLAMs) located at the central office or remote distribution points.

(4) **Subscriber** - Customer delivery of IPTV is provided over the existing loop plant and the phone lines to homes using the higher-speed DSL technologies such as ADSL2+ and VDSL based on subscriber’s package and service subscription. Service providers may use a combination of Fiber-to-the-Curb (FTTC) and DSL technologies or implement direct Fiber-to-the-Home (FTTH) access depending on the richness of their IPTV service offerings.

(5) **IPTV Client** - The IPTV Client is the functional unit, which terminates the IPTV traffic at the customer premises. This is a device, such as a set-top box, that performs the functional processing, which includes setting up the connection and QoS with the Service Node, decoding the video streams, channel change functionality, user display control, and connections to user appliances such as a standard-definition TV or HDTV monitors.

(6) **Customer Premises Equipment (CPE)** - In the IPTV context, the CPE device located at the customer premise provides the broadband network termination (B-NT) functionality at a minimum, and may include other integrated functions such as routing gateway, set-top box and home networking capabilities.

II. **IPTV Layered Architecture**

IPTV systems communication network has multiple layers ranging from a base layer that physically transports data to the layer that presents the media to the viewer. Each layer not only performs a specific function as per its defined protocols, but also receives services from the protocol layer below it and provides services to the protocol layer above it.

![IPTV Layers](image)

**Fig. 2 IPTV Layers**

(1) **Physical Layer** - The base layer ‘Physical Layer’ is responsible for converting bits of information into data packets that are transferred on a network. Each IPTV client has a Mac address associated with it. Hence the responsibility of MAC layer is for
requesting and managing access to the physical channel. The main job of Internet protocol layer is adding the network address to packets so they can travel across the network to reach their destination. DHCP server assigns a unique IP address to each IPTV client using DHCP protocol. It is a network protocol that is used to configure network devices so that they can communicate on an IP network. DHCP assigns unique IP addresses to devices.

(2) Transport Layer – Transport layer is responsible for transferring packets between the source and the destination over RTP/UDP transport mediums. RTP is designed for multicast and unicast of real-time data. UDP uses a simple transmission model without implicit handshaking. UDP provides an unreliable service and datagrams may arrive out of order, duplicate or miss without notice.

(3) Session Layer: The session layer manages and controls the transfer of the media components for the program channel using video encoding techniques. The Moving Pictures Experts Group, or MPEG, is the body responsible for the standards that we often use for video encoding. In IPTV MPEG2, MPEG2 and H264 standards are used for video encoding.

(4) Pack elementary stream – Packet elementary stream is a specification which is responsible for carrying the media components in packets to the transport streams. The media streams are transformed into packets by encapsulating sequential data bytes from the elementary stream inside PES packet headers.

(5) Application Layer: The application layer provides the interface between the Setup Box and the end user using UI graphics screens on TV.

III. APPLICATIONS AND SERVICES OFFERED BY IPTV

Few key features for initial IPTV deployment are delivery of subscribed live TV channels or stored video. These applications enable service providers to initiate the “triple play” – video, voice and data.

- Digital Broadcast TV
- Video on Demand (VoD)

Digital Broadcast TV - Digital Television (DTV) is a latest broadcasting technology that has advanced your television viewing experience. DTV has enabled media providers to offer television with quality picture and sound. The advent of higher-speed DSL technology such as ADSL2, ADSL2+ and VDSL, enables IPTV as a compelling and competitive alternative. IPTV is currently in testing, or planning, stages with a number of telecom service providers in North America, Europe and Asia.

It also offers multiple programming choices, called multicasting and interactive capabilities. IPTV enables more content variety with a larger number of channels. The multiple channels that can be included in the digital signal mean that more programs will be available.

Video on Demand (VoD) – Video on Demand (VOD) are systems which allow users to select and watch video content on demand. When the subscriber selects a movie to view, an RTSP (Real-time Streaming Protocol) connection to the video content server is established, initiating the playback (streaming) to the set-top box (STB) which decodes and finally plays the video back.

The following on-Demand services are supported

- Video on Demand (VoD) for renting movies
- Subscription for VoD (SVoD) for renting VoD content without paying an extra fee
- Free on Demand (FoD), similar to SVoD but with item assignment instead of category assignment
- Other compelling IPTV applications and potential revenue-generating services, which can be enabled once the initial IPTV infra-structure is in place, are:
  - Video telephony and Video conferencing
  - Remote Education
  - Home Security/Monitoring Cameras
  - Voice over IP
  - Messaging/Recommendations
  - Chat around content
IV. WHY TO SELECT THE IPTV SERVICES FOR HOME DISTRIBUTION

Stream Quality – Depending on the video quality and parameters used, an IPTV service provider can deliver the following type of streams to a typical household:
Up to 3 Mbit/s to 4 Mbit/s bandwidth standard broadcast quality TV (SDTV) streams
Up to 9 Mbit/s to 10 Mbit/s bandwidth additional high definitions TV (HDTV) stream

Fig. 3 Multicasting

Channel Multicast - Internet Protocol (IP) multicast is a bandwidth-conserving mechanism for reducing data network traffic by simultaneously delivering a single stream of information to thousands of recipients. In the context of the data network, the distribution to all subscribers is achieved by multicast implementations in the core and the access networks. For example, if 15 IPTV channels are broadcast and each channel is encoded by H.264 codec providing a gross bit rate of 2 Mbit/s (incl. Ethernet overhead), 30 Mbit/s bandwidth is required for the IPTV service. The calculated 30 Mbit/s IPTV traffic will be transmitted via the network operator’s IP core network to the DSLAMs independent of the number of end customers. This amount of traffic does not affect the throughput of the IP core network dramatically.

Easy installation – This is an important point to consider that most IPTV end users are not ‘technical’ and IPTV providers may desire to eliminate or minimize service calls to the customer premises. Refer to the diagram above, where the xDSL modem/routing gateway is installed at one location and the set-top box/IP client is installed on the network at different locations throughout the home. In this scenario, the installer would have to test to see if the existing home network meets the quality of service requirements for IPTV service. Otherwise, a new home network and its associated wiring/cabling may need to be installed prior to the IPTV service installation. Obviously, to expedite the installation process, it is most desirable not to have to install new wiring/cabling and to use the existing physical infrastructure in the home.

As IPTV matures, equipment vendors will provide higher levels of integration by combining more functionality into single easy-to-install boxes. Similarly, the home networking technology used must allow home users to self-install the network. The service providers may offer remote network management and diagnostics/support functions in their customer premises equipment, which may interface with the home network technology to provide a more comprehensive IPTV customer-care.

Privacy and Security - The home network must be a closed, secure network in which access is limited to users and devices within the home. Software-based content protection and public key infrastructure (PKI) encryption solution is used for VOD and broadcast streams. It is designed to be infinitely scalable and fully redundant. STB client for certificate management and the decryption of VOD and broadcast streams.

Service Quality – The wide area networks used by the service providers have mechanisms that control the service quality. IPTV is a real-time service that has very stringent QoS requirements, specifically packet loss and delay. A small amount of delay does not directly affect the quality of experience of IPTV. However, a delay longer than 1 second may result in a less than satisfying end user experience if a neighbour with traditional TV service is able to watch a goal in an important soccer game.
several seconds in advance. The VQE solution is designed to improve the quality of IPTV services and the subscriber’s viewing experiences. As video is less tolerant of network factors like jitter, delay or packet loss, additional functionality is requested to improve the quality of the IPTV service in order to meet the expectations of the subscriber. The VQE solution provides the following functionality:

Selective Retransmission (RET).
Forward Error Correction (FEC).
Rapid Channel Change (RCC)

The VQE client (VQE-C) on the STB requests a server in the network to selectively retransmit a packet that is either lost or corrupted and can’t be repaired by other means (e.g. FEC). This solution requires an additional server (VQE Server, VQE-S) that stores the multicast packets for each TV channel for a certain amount of time in order to be prepared to serve unicast requests that are received from the STBs attached.

Reliability – IPTV technology enables service providers to offer more content and channels with more viewing choices for the users. With declining prices on TVs and flat panel displays, it is conceivable that users may wish to install additional TVs/displays at many additional locations in the home – bedrooms, recreation/living room, kitchen, office/den, basement etc. This allows users to conveniently watch digital television and movies on demand in any room or area of the house.

It is important that the home network be easily expandable and scalable to reach all areas of the house.

This network design assumes that the multicast replication is achieved in the DSLAM, and that that and the multicast traffic is distributed from the redundant IP edge via a multicast VLAN to the DSLAMs. Usually there is only one IPTV edge location in the network. In some situations, geographic redundancy is required and there are two IPTV edge locations. However, there are usually multiple IP edges, one for each regional access and aggregation network. For the access and aggregation network, two options are generally possible. First, the topology can be built as a tree structure, using Rapid Spanning Tree Protocol (RSTP) or Multiple STP (MSTP) to resolve redundant paths. Secondly, ring topologies are emerging to improve reliability. The innovative Nokia Siemens Networks Ethernet Ring Protection (ERP) provides very fast switch-over times in case of ring failures.

V. THE SET-TOP-BOX BOOT-UP PROCESS

When the STB box boots the first task it performs is a DHCP request.

That STB is purely dependent upon successfully contacting the DHCP server and also receiving the proper parameters from that server. Once the STB has successfully obtained a DHCP lease, it uses the information to perform the following actions:

• Set its time
• Download software and OS (during first boot)
• Listen to the SAP announcements from the MMF server
• Basic STB Boot order

The basic boot sequence includes:

I. Requests DHCP.
II. Sets time using RDATE.
III. Establishes connectivity with NTP server.
IV. Joins SAP announcement channel.
V. Downloads the following documents:

GLOBALINSTALL.xml
DTVLINEUP.xml
PIPEDSCHEDULE.xml
The authentication process is based on a “User Token” that is generated as a combination of three inputs namely:

- Set Top Box ID (specific to an STB)
- Username
- Password

Set Top Box Details and working
- Operating System: Suse Linux
- Memory: 128 Mb available
- On each action that has been initiated from the remote controller, there is a corresponding HTTP request that STB (Set Top Box) generates to the EPG (Electronic Program Guide) server, which thereby replies or forwards to the respective server.

- Each remote button is having a separate identification number that is passed on to the STB’s API (Application Program interfaces), which converts the request into the corresponding HTTP request.

- The response from the server is loaded on the browser.

VI. THE REQUIREMENTS OF THE BILLING, OSS AND BSS PLATFORMS [7]

- Business Support Systems (BSS)
- Operation Support Systems (OSS)

This solution provides a full range of customer care and billing capabilities including:

- Billing collection:-

Collection of billing records for voice and xPlay networks.

Vendor and product name(s): Alcatel 8965 C3S.

- Billing and Customer Care:-

Integration of billing, customer care management, rating and invoicing systems.

Vendor and product name(s): Alcatel convergent billing powered by LHS – BSCS 6.

- Contact centre:-

Integration of contact centre applications.

Vendor and product name(s): Genesys Suite

VII. LIST OF SECURITY THREATS RELATED TO IP NETWORKS [11][12]

Digital Content Provider - This component requires security technology that can prevent illegal use of digital content by controlling the ability to access the content and allowing access only to authorized users

Service Provider - IPTV services should have the capability to identify legitimate subscribers in order to prevent illegitimate access. Robust authentication and strict authorization schemes are required to prevent illegitimate subscribers from accessing service networks and servers. Also, IPTV services should be able to limit the
ability of legitimate subscribers to access content. (e.g. limit the number of times the same user can access the network and prevent uploading and downloading of unallowable data), in order to prevent abuse of network resources.

**Network Provider** - In this component, technology related to multicast or protocol security is required for secure network transmission. In particular, robust authentication is required for data transmission or access requests.

**Terminal** - Preventive measures should be developed to control illegitimate use of digital content previously delivered to users and the release of subscriber information.

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DETAILED REVIEW OF WAVELET TRANSFORM APPROACH FOR P300 BASED BCI SYSTEM

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Abstract—This paper discusses and emphasizes on the use of wavelets in P300 based BCI systems. Electroencephalograph (EEG) signal processing involves feature extraction that is one of the key issues in P300 based brain-computer interface systems (BCI). Various techniques have been used to extract features from P300 signals for classification purposes. The most common techniques are Fourier Transform, Short Term Fourier Transform, and Wavelet Transform etc. Since EEG is a non-stationary in nature, researches have shown that wavelet transform is best suited to represent time-frequency information of ERP signals, specially the P300. The goal of this study is to investigate the analysis of the P300 features extracted through wavelet transform for P300 based BCI systems. The paper mentioned the limitations of wavelet based BCI systems and compared the various wavelet methods.

Keywords—P300, EEG (Electro-Encephalogram), Wavelet Transform, Interface, Brain-Computer Interface (BCI).

INTRODUCTION

Recently there has been a growing interest in EEG based Brain Computer Interface Systems (BCIs) which are extremely useful for unblessed people described in [1] [2]. The electroencephalogram (EEG) is the brain activity which if recorded with respect to a particular stimulus known as evoked related potential (ERP). The well-known and most popular type of EEG’s neurological phenomenon is P300, generated and obtained from the central-parietal region of the brain in response to rare or unexpected events [3]. Amplitude and latency are the characteristics of ERP signals. The time evolution of the amplitudes of such non-stationary signals does not allow the accurate retrieval of the frequency information of the signal. The signals acquired are treated as raw signals which reflect only time-domain information i.e. Time-Amplitude representation. This does not contain frequency information. For getting the frequency content of the signal, the transformations are required i.e. Frequency-Amplitude representation. Therefore, the transformations of the signal are required to get the frequency information. The common techniques are Fourier Transform (FT), Short-Time-Fourier Transform (STFT), Wavelet Transform (WT); include Continuous Wavelet Transform (CWT) and Discrete Wavelet Transform (DWT). Fourier Transform (FT) is one of the most known techniques. On applying Fourier Transform, it provides only frequency domain information of the signals, no time domain information. It just gives the information of each frequency that exists in the signal, but it does not tell when in time these frequency components exist. Therefore, it is merely used for the signals whose frequency content does not change in time called stationary signals [4] [5]. J. Fourier introduced Fourier Transform as transformation technique that showed any periodic function as an infinite sum of periodic complex exponential function defined by the following two equations [5]:

\[ X(f) = \int_{-\infty}^{\infty} x(t) e^{-2\pi j ft} dt \]

\[ x(t) = \int_{-\infty}^{\infty} X(f) e^{2\pi j ft} df \]

where x denotes signal, t denotes time, f denotes frequency.

\( X(f) \) denotes signal in frequency domain i.e. FT of \( x(t) \) \( x(t) \) denotes signal in time domain i.e. inverse FT of \( X(f) \) integral correspond to all time instances and treated as window function of FT.
If the value of integration result is

<table>
<thead>
<tr>
<th>Large</th>
<th>Small</th>
<th>Zero</th>
</tr>
</thead>
<tbody>
<tr>
<td>Signal x(t) contain high amplitude component of the frequency f.</td>
<td>Signal x(t) contain small amplitude component of the frequency f.</td>
<td>Signal x(t) does not contain the frequency f.</td>
</tr>
</tbody>
</table>

The major disadvantage of Fourier analysis is the lost of brief information in the frequency domain. Hence, cannot be used for the signals whose frequency content varies over the time called non-stationary signals. Almost, all biological signals like Electroencephalography (EEG), Electromyography (EMG), and Electrocardiography (ECG) etc. are non-stationary in nature. In such signals and an event-related potential (ERP) signals, the Time-Frequency information is needed. The most common ERP is P300 signal that is the response of the brain to a specific stimulus like flashlight, occurrence of rare or surprising task-relevant stimuli [6]. For such non-stationary signals, the frequency domain is typically dependent on time domain. If some portion of a non-stationary signal is stationary, then the Short-Time-Fourier Transform (STFT) is applied. In this technique, the signal is separated into small portions and a window function ‘w’ is chosen. The width of this window function ‘w’ is equal to the portion of the signal that is stationary. Different slide windows exist like the Hamming, Bartlett, or Kaiser windows [6]. The window function ‘w’ is applied to the signal and somewhere it will overlap with the stationary portion of the signal. When these two are overlapped, apply Fourier transform on the product of window function ‘w’ and signal, which produces another signal. Then this window function ‘w’ is shifted and the process is repeated until the end of the signal. Therefore, STFT is windowed version of the FT that gives different FT’s of frequencies at different times or STFT= FT (signal* w).

The shortcoming of STFT is the resolution problem of the width of the window function. A time window is chosen and applied, same for all frequencies of the signal, due to which important information can be lost at very low or high frequencies. The difference between Fourier Transform and Short-Time-Fourier Transform (STFT) is in former window function ‘w’ is infinite therefore, wide window is chosen that gives good frequency resolution but poor time resolution where in later window function ‘w’ is finite therefore, narrow window is chosen, to obtain stationary signal in non-stationary signal, that gives good time resolution but poor frequency resolution. The information provided by STFT is limited by the fixed size window that causes the frequency resolution to get poorer. Therefore, wavelet transform (WT) provides solution to this resolution problem in which choosing the window function ‘w’, once and all in the entire signal analysis, is application dependent. In WT, the time-frequency window known as mother wavelet that is flexible due to which important information cannot be lost at very low or high frequencies. It entails multi-resolution analysis (MRA) that analyzes the signal at different frequencies with different resolutions. The window functions ‘w’ are chosen and derived from the main mother wavelet window using translation and dilation operations. Translation means shifting ‘w’ along time axis and dilation means scaling, compressing or stretching the mother wavelet. WT allows the use of the long time windows for producing good frequency localization at low frequency, and short time windows for producing good time localization at high frequency [7]. The advantage of WT is the size of window that changes as the transform is computed. Unlike STFT, which has a constant resolution at all times and frequency, the WT gives a variable resolution. Therefore, WT is best EEG signal analysis tool which provide time, frequency and amplitude information. The continuous wavelet transform an alternative to STFT, is a form of wavelet transform that developed to overcome the resolution problem. Later, in 1976 Croiser, Esteban and Galand devised a technique to decompose time signals, known as discrete wavelet transform (DWT). The basic difference between Continuous
wavelet transform and Discrete wavelet transform is depicting in Table 1 [5] [8].

**Table 1: Difference between CWT and DWT**

<table>
<thead>
<tr>
<th>CWT (Continuous Wavelet Transform)</th>
<th>DWT (Discrete Wavelet Transform)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Based on mother wavelet or the basis function</td>
<td>Based on sub-band coding</td>
</tr>
<tr>
<td>For transformation the wavelet functions are derived from the mother wavelet</td>
<td>Digital filtering techniques are used to yield a time-scale representation of the digital signal</td>
</tr>
<tr>
<td>Involves translation (shifting) and Scaling (dilation or compression)</td>
<td>The signal is passed through filters with different cutoff frequencies at different scales</td>
</tr>
<tr>
<td>CWT is computed by the convolution operation of the signal and the basis function</td>
<td>DWT is computed by successive low-pass and high-pass filtering</td>
</tr>
<tr>
<td>The computation may consume significant amount of time and resources, depending on the resolution required</td>
<td>Easy to implement and reduces the computation time and resources required, therefore a fast computation of Wavelet Transform</td>
</tr>
</tbody>
</table>

This paper focuses on the research done by various authors using wavelet transform for feature extraction. The paper is divided into the following sections. In Section II, compares various wavelet methods for existing P300 based BCI systems. The section III discusses the limitations of wavelet based P300 BCI Systems. Finally, conclusion comes in Section IV.

**WAVELET TRANSFORM (WT) METHODS FOR P300 BASED BCI SYSTEMS**

The P300 signal pattern can describe by specifying the set of time domain features, frequency domain features and time-frequency domain features. Due to the transient nature of P300 signal, time-frequency features are suitable. To obtain such features the wavelet transform is the best signal analysis tool. Wavelet Transform gives the best interpretation of the brain signal in terms of time and frequency. The last 14 years survey of using wavelet transform for P300 based BCI systems have already been discussed in [9]. This section further explores the survey of various wavelets used.

In 1998, the Quadratic B-spline wavelet transform (WT) was used to analyze the functional components of P300 ERP. The study discussed several frequency ranges of P300 i.e. delta, theta and alpha frequency in oddball paradigm. The analysis of time-frequency components were performed by repeated measures ANOVA. The stepwise discriminant analysis (SWDA) was used to differentiate targets and non-targets, which showed 78% of the targets and 81% of the non-targets. The research showed that when oddball stimuli are very uncommon then short latency and frontal predominance of P3a can be obtained [10].

In 2006, a deception detection method was investigated using wavelet. Here, the features were extracted through quadratic B-spline wavelets that differentiated deceptive and truthful responses. The signals were evaluated up to 6th scale i.e. d4 (~15–31 Hz) roughly corresponds to beta, d5 (~7.5–15 Hz) roughly corresponds to alpha and d6 (~3.75–7.5 Hz) roughly corresponds to theta. The d5 and d6 failed to show any significant difference between ‘target’ and ‘truth’ cards. But in d4, 10 out of the 31 coefficients that are d4_8, d4_10, d4_16, d4_17, d4_20, d4_21, d4_24, d4_27, d4_29, d4_31 showed statistically significant differences. The results showed that joint time-frequency EEG features extracted through wavelet analysis might provide a more reliable method for detecting deception than standard ERPs [11].

Another method was introduced in 2006 for analyzing the P300 component while performing a cognitive task using wavelet. Daubechies wavelet of DWT was used to detect the presence of P300 in individual trials. In addition, the wavelet filtering was computed from grand-average of wavelet coefficients to remove noise and unwanted frequency components[12].

In 2007, an algorithm was proposed for the P300 detection using single trial. The two estimation models were used for detection i.e. Kalman
filtering and Daubechies-4 wavelet as Wavelet Transform. The discrimination analysis was performed by PCA. The three methods were used in the discrimination analysis; linear discriminant analysis, Mahalanobis distance, and Bayes’ theorem. It had been found that the linear discriminant analysis method can perform discrimination best in the accuracy of 97.13% with Kalman filtering and 96.04% with Daubechies-4 wavelet among three discrimination analysis methods [13].

In 2009, the Mind Speller, a Brain-Computer Interface application was developed for disabled patients. The data were filtered with a 4th order zero-phase digital Butterworth filter and properly cut into signal epochs. Three types of features extracted were down sampled signals, CWT and Common Spatial Pattern. The Least-Squares Support Vector Machine used for classification. The results showed that subjects were able to communicate a character in less than ten seconds with an accuracy of 94.5% [14].

In 2009, wavelet transforms were used for recognizing and classifying P300 signals. The oscillatory transient features were detected and the distinct functional components of P300 were extracted. The db4, bior2.4, bior4.4, bior5.5, coif2, sym4, and sym6 are the most useful types of DWTs. In this work, 20-30 coefficients were extracted using these wavelets. A five-octave wavelet transform was performed which produced five sets of coefficients in the 60-120 Hz, 30-60 Hz (gamma), 1530 Hz (beta) , 8-15 Hz (alpha) and 4-7 Hz (theta) frequency ranges and the residues in the 0.5-4 Hz frequency band. The feature analysis was limited within a 20 by 10 matrix. A t-test was performed to choose the prominent features with respect to the P300 wave. The threshold at 70% of the total energy was performed which reduced number of features from 200 to about 20-30. The quadratic discriminant analysis performed as classification. The 100% correct decisions obtained with few repetitions. Further experiments on different data sets are required. In addition, this method can be used for an online BCI [15].

A P300-based Guilty Knowledge Test (GKT) was developed in 2009 as a new method for the psycho-physiological in which three types of features were extracted. The first type was Morphological features like Latency, Amplitude, Latency/amplitude ratio, Absolute amplitude, Absolute latency/amplitude ratio, Positive area, Negative area, Total area, Absolute total area, Total absolute area, Average absolute signal slope, Peak-to-peak, Peak-to-peak time window, Peak-to-peak slope, Zero crossings, Zero crossings density, Slope sign alterations. Second type was Frequency features like mode frequency, median frequency and mean frequency. Finally, the Wavelet features which were extracted using Quadratic B-Spline functions, used as mother wavelets due to similarity with the evoked responses. The features extracted were evaluated using statistical method i.e. t-test and selected using genetic algorithm (GA). The linear discriminant analysis (LDA) was used for classification of innocent and guilty subjects. The correct detection rate was 86% [16].

In 2010, a method was described for the detection of P-300 rhythm using Two-level Discrete Wavelet Filter Bank i.e. the discrete wavelet transforms (DWT). The de-noising and blind source separation was performed by Independent Component Analysis algorithm. DWT outperforms the others as an analyzing tool for P300 rhythm detection in comparison with the Short Time Fourier Transform (STFT) and Wigner–Ville Distribution (WVD)[17].

Another algorithm was proposed in 2010 for P300 feature detection method based on wavelet transform and Fisher distance. Here, first wavelet transform was applied to EEG signals and then Fisher distance had been calculated in order to analyze the divisibility of the feature to obtain the optimal features. The neural network was used as classifier to classify the selected features. This proposed method increased the
classification accuracy by 1.2% and decreased classification time by 73.5% [18].

In 2011, responses to auditory stimulation in the ‘alpha’ range (8 Hz-13 Hz) in the low and high education groups were studied using the Wavelet Transform (WT). Auditory evoked responses of total 16 healthy subjects (8 in each group, low and high education groups) were studied with 2 different auditory stimuli (standard and target stimuli). Upon the stimulus types, event-related responses in the 8 Hz-13 Hz (‘alpha’) ranges were distributed mainly in the central (Fz, Cz and Pz) locations. The Low education group had significantly high alpha power at Fz and Cz locations comparing with the High education group in the case of standard stimuli, whereas remarkable increased alpha power was found in the same group in the case of target stimuli at Pz area. It is concluded that the distributed origin of event-related alpha oscillations had the relationship with educational experience [19].

A P-300 rhythm detection system was developed in 2011 using an Adaptive Neuro Fuzzy algorithm (ANFIS). The de-noising and blind source separation was performed using an Independent Component Analysis algorithm (ICA). The features i.e. P300 rhythms were extracted using Daubechies-4 wavelet i.e. the discrete wavelet transform (DWT) which then fed to the ANFIS system. Experimental results showed 85% detection of the P300 rhythm [20].

In 2012, the work focused on selecting minimal set of effective features for detecting P300 component. The discrete Daubechies4 (db4) wavelet was used to obtain coefficients. For selecting minimal channels, the Bhattacharyya distance was used in decreasing order and binary particle swarm optimization algorithm (IBSPO) was used to select channels that are more effective. This reduced the search space and processing time of the IBSPO algorithm. The preprocessing was performed using windsorizing method to remove the outliers. In 15 and 5 trials of dataset IIb of BCI competition 2005, the accuracy result was 97.5% and 74.5% respectively, using Bayesian linear discriminant analysis (BLDA) [21].

This section concludes that the wavelet analysis can effectively used to extract the joint time-frequency P300 features in the proposed method. Moreover, the wavelet transform also helps in removing the noisy and non-meaningful information from ERP signals. Therefore, it is clear that for ERPs particularly P300, wavelet analysis is a successful feature extraction method [22].

**MAJOR PROBLEMS IN WAVELET BASED P300 BASED BCI SPELLER SYSTEMS**

The survey on WT for P300 based BCI systems showed that there are still some gaps in using such techniques like need of an adaptive filter, the data precision problems in processing the P300 estimation, require experiments with more data sets and the optimal electrodes for channel selection and the set of optimal DWT for sub-band selection were subject dependent. In addition, the tuning of the classifier is very time consuming. The data precision problems have reported in processing the P300 estimation, the feature extraction, the classification and the discrimination remain in the future [13]. The study reported that after the time lapse, the experiment became tedious for most of the users, with the consequence of generating low-level P300 signals, undetectable in this experiment [17]. The experiment became tedious which result in generating low-level P300 signals, reported in [20]. The results of [21] proved that the set of optimal electrodes for channel selection and the set of optimal DWT for sub-band selection were subject dependent.

A major technical issue in the P300-based BCI speller reported the increase in accuracy at a high computational overhead of both during training and execution [23]. Another view also commented about the time consuming training of the LSSVM classifier [24].

Another BCI system [25], described the low divisibility of the extracted features, which increased the training and testing time of
classifier. The P300 based BCI speller systems developed using hybrid P300 and motor imagery signals, reported the reduced accuracy and slow because the user has to wait for randomly generated highlight to reach the letter under focus [26].

The mentioned P300 based speller exhibit the maximum accurate results with near 100% classification but with only at 4 trials and minimum 70% accuracy in terms of time taken to spell all letters.

The limitations of various P300 based BCI Speller system and the semi-supervised learning approach for feature discovery motivated to suggest the proposed method in Section VI.

**CONCLUSION**

The aim of this analysis is to explore the advantages and ability of wavelet features that are highly suited for brain signal analysis. The basic idea of using wavelets is to improve the precision and speed of the BCI system. We showed a survey of existing P300 based Brain computer Interface Systems and discussed the use of wavelet transform for feature extraction in P300-based BCI systems. In such systems, the distinct functional components of P300 are extracted. The wavelet features that are short-lived in a signal are detected and extracted, using various methods like Daubechies 4 wavelet, Kalman filtering, Quadratic B-spline etc, to form feature vectors for classification purposes as shown in Figure 1. Based on the research results, the Daubechies 4 wavelet have outperform well as compared to other techniques. The Daubechies 4 wavelet decomposes the signal at different frequency bands, with different temporal resolution [4].

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**REFERENCES**


A REVIEW OF ROUTING PROTOCOLS OF MANET

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ABSTRACT: Mobile means random and perhaps constantly changing or moving and ad-hoc means for this or temporary without any fixed infrastructure. Mobile ad-hoc network is a collection of mobile nodes forming an instant network characterized by wireless links, changing topology, low transmission power and asymmetric line. The wireless links in this network are highly prone to error. Routing is a challenging task for networks that do not offer centralized infrastructure, like in ad-hoc wireless networks that offer unrestricted mobility. Based on Routing Update Mechanism, there are two broad categories of Routing Protocol viz. Table-Driven Routing Protocols includes: protocols like DSDV & WRP and On-Demand Routing Protocols include protocols like DSR, AODV & LMR Where, Zone Routing Protocol (ZRP) combines advantages of the proactive and reactive approaches by maintaining an up-to-date topological map of a zone centered on each node. In this paper, an attempt has been made to analyze three well known protocols namely AODV, DSR, DSDV, ZRP.

Keywords: Ad Hoc Networks, DSDV, DSR, AODV, ZRP, Comparison between different Routing protocols.

I. INTRODUCTION
Unlike traditional networks in mobile ad-hoc network (MANET) all nodes can be mobile while communication and any node can disappear or join anytime at any location in the network. It is an infrastructure less having no base station. Mobile ad-hoc network is characterized by dynamic topology having low power consumption & bandwidth. The nodes which are in the transmission range of each other can communicate directly otherwise communication is done through intermediate nodes.

The wireless network can be classified into two types: infrastructure and infrastructure less network. In infrastructure networks, the mobile node can communicate with each other as the base stations are fixed and if the nodes get out of range of a base station, it gets into the range of another base station as the nodes are mobile[1]. In infrastructure less network the node is mobile while communicating, as there is no fixed base station and all the nodes in the network act as routers. The mobile node in the network dynamically established routing among themselves to form their own networks [1]. Some of the challenges of MANET are Limited bandwidth, battery constraints, routing overhead, asymmetric link, speed, scalability, packet loss and quality of services.

We have made an attempt to compare and analyze on some performance parameters on our own network scenario. We consider our own network case where 9 nodes are placed randomly.

II. ROUTING PROTOCOLS
Routing is the process of selecting paths in a network along which to send network traffic [2]. Routing in ad-hoc network is different then wired network due to mobility of the nodes. Routing protocols are basically classified as following:

Fig. 1. Classification of Routing Protocol

A. Table Driven Routing Protocol
These protocols are also known as proactive routing protocol. Each and every node maintains information about every other node [3]. As,
Routing information is maintained in routing table and is updated time to time as the network topology changes. Some of the exiting table-driven or proactive protocols are: - DSDV (destination sequence distance vector), WRP (wireless routing protocol), GSR (global state routing), STAR (source tree adaptive routing), DREAM (distance routing effect algorithm for mobility) and OLSR (optimized link state routing protocol).

B. On-Demand Routing Protocol

These protocols are also known as reactive routing protocols, routes are created in an on-demand manner. When transmission occurs from source to destination, it invokes route discovery procedure [1]. Source node sees its route cache for the suitable path from source to destination if the route is available then it use that route to send data packet otherwise it initiate route discovery process.

Some of the existing on-demand or reactive routing protocols are: DSR (Dynamic Source Routing), LMR (Lightweight Mobile Protocol), TORA (Temporally Ordered Routing Protocol) & ABR (associativity-based routing).

C. Hybrid Routing Protocol

Hybrid protocol is presented to overcome the shortcoming of both table-driven and on-demand routing protocols [4]. It uses the route discovery mechanism of reactive routing protocol and table-maintenance mechanism of proactive routing protocol. So, as to avoid latency & overhead problem. Some of the existing hybrid routing protocols are: - ZRP (zone routing protocol) and IARP (intra-zone routing protocol).

III. WORKING of TABLE-DRIVEN ROUTING PROTOCOL

Destination sequenced distance vector (DSDV) protocols is based on bellman-ford shortest path algorithm. Each node has a table, which contains the shortest path to every other node in the network. These tables are being updated and forwarded to other nodes in the network whenever a change is detected. When a node receives an update it can either update the tables or hold it for a while in order to select shortest route [4].

Periodically when network topology changes are detected, each mobile node sends routing information using broadcasting or multicasting a routing table update packet. The update packet starts out with a metric of one to direct connected nodes. This indicates that each receiving neighbor is only a hop away from the node. It is different from that of the conventional routing algorithms. After receiving the update packet, the neighbors update their routing table with incrementing the metric by one and retransmit the update packet to the corresponding neighbors of each of them. The process will be repeated until all the nodes in the ad hoc network have received a copy of an updated packet with a corresponding metric. The update data is also kept for a while to wait for the arrival of the best route for each particular destination node in each node before updating its routing table and retransmitting the update packet [6]. If a node receives multiple update packets for a same destination during the waiting time period, the routes with more recent sequence numbers are always preferred. Fig. 2 illustrates the example of DSDV routing protocol.

Fig. 2: An example of the DSDV

<table>
<thead>
<tr>
<th>Destination</th>
<th>Next</th>
<th>Metric</th>
<th>Seq.no.</th>
<th>Install Time</th>
<th>Stable Data</th>
</tr>
</thead>
<tbody>
<tr>
<td>A</td>
<td>A</td>
<td>0</td>
<td>A-300</td>
<td>001000</td>
<td>Pri_A</td>
</tr>
<tr>
<td>B</td>
<td>B</td>
<td>1</td>
<td>B-100</td>
<td>001200</td>
<td>Pri_B</td>
</tr>
<tr>
<td>C</td>
<td>B</td>
<td>2</td>
<td>C-550</td>
<td>001200</td>
<td>Pri_C</td>
</tr>
<tr>
<td>D</td>
<td>B</td>
<td>3</td>
<td>D-322</td>
<td>001200</td>
<td>Pri_D</td>
</tr>
<tr>
<td>E</td>
<td>E</td>
<td>1</td>
<td>E-100</td>
<td>001200</td>
<td>Pri_E</td>
</tr>
<tr>
<td>F</td>
<td>E</td>
<td>2</td>
<td>F-200</td>
<td>001200</td>
<td>Pri_F</td>
</tr>
<tr>
<td>G</td>
<td>E</td>
<td>3</td>
<td>G-280</td>
<td>001200</td>
<td>Pri_G</td>
</tr>
<tr>
<td>H</td>
<td>E</td>
<td>4</td>
<td>H-300</td>
<td>001200</td>
<td>Pri_H</td>
</tr>
</tbody>
</table>

Fig. 3: Table Entries of DSDV Protocol

Fig. 3 represents the table entries of DSDV protocol.
Under this we have sequence number, install time and stable data which is defined as:

- **Sequence Number**: It originated from destination. Ensures loop freeness.
- **Install Time**: when entry was made (used to delete stale entries from table).
- **Stable Data**: pointer to a table holding information on how stable a route is. Used to damp fluctuations in network.

**IV. WORKING OF ON-DEMAND ROUTING PROTOCOL**

**A. DSR (Dynamic Source Routing)**

DSR is an on-demand/reactive routing protocol. It uses source routing. In source routing only source provides information regarding whole path; intermediate node does not provide any information about destination. If any source contains more than one path in its Cache, which path to choose will entirely depend on the source.

DSR works in two parts Route Discovery and Route Maintenance. Whenever a node finds a new path towards destination, it stores that path in its Cache for future use.

**i) Route Discovery Process**

When a source is ready to send data packet to destination D it put source route in the header of the packet. Here source route is a sequence of hop between source and destination. So, source node S first searches in its route Cache. If it doesn’t found any route to destination D in its Cache than it start route discovery. At starting of route discovery Source S send a Route Request (R.REQ) with source address, destination address and ID attached with the request [12]. Any intermediate node checks for ID, its address in route record. If found then simply discard this packet otherwise append its address in its route record. When finally this R.REQ reaches at destination node D it responds this query with Route Reply (R.REP) to source node S. At destination D there is more than one R.REQ to propagate from different path; they reply all R.REQ by R.REP [9]. So, as result of single route discovery a node can learn multiple routes. For example we take our own network scenario for Route Request in DSR protocol.

**ii) Route Maintenance Process**

As we know that due to the mobility and high interference in wireless network, life time of link between two nodes no longer exists. It may be possible that link present now must not be working in future. For this purpose Route Maintenance procedure is introduced. Whenever an intermediate node finds a broken link in between the path from source S to destination D it sends a Route Error (R.ERR) message back to the source S [12]. When this Route Error message arrives at source it remove that link from its Cache and find another route for...
sending data of the specific destination [10-13]. If there is no route found in Cache of specific destination then Route Discovery process is initiated. Fig. 6 represent Route Maintenance phase in DSR. We take our own network scenario.

**B. AODV (ADHOC ON-DEMAND DISTANCE VECTOR ROUTING PROTOCOL)**

AODV is also a reactive routing protocol. It is also a variation of Destination-Sequenced Distance Vector (DSDV) routing protocol which is collectively based on DSDV and DSR. In AODV, routes are not maintained from each node to every other node in the network rather they are discovered as and when needed and are maintained only as long as they are required. Major difference between AODV and DSR is that DSR uses source routing in which a data packet carries the complete path to be traversed, whereas in AODV, the source node and the intermediate nodes store the next-hop information corresponding to each flow for data packet transmission [13].

**A. Routing Discovery**

When a route is not available for the destination, a route request packet (R.REQ) is flooded throughout the network. The Route Request contains the following fields:

![Route Request Format](image)

Fig. 7: Format of Route Request

The request id is incremented each time the source node sends a new R.REQ, so the pair (source address, request id) identifies a R.REQ uniquely [15]. On receiving the Route Request each node checks the source address and request id. If the node has already received the Request from the same source with same request id the packet will be discarded. Otherwise the R.REQ will be either forwarded or replied with a R.REP message: if the node has no route entry for the destination, or it has one but this is no more an up-to-date route, the R.REQ will be rebroadcasted with incremented hop count and if the node has a route with a sequence number greater than or equal to that of R.REQ, a R.REP message will be generated and sent back to the source [13]. Every R.REQ carries a time to live (TTL) value that specifies the number of times this message should be re-broadcasted. This value is set to a predefined value at the first transmission and increased at retransmissions. Retransmission occurs if no replies are received [14, 15].

If a node is the destination, or has a valid route to the destination, it unicasts a Route Reply message (R.REP) back to the source. This message has the following format:

![Route Reply Format](image)

Fig. 8: Format of Route Reply

Fig. 7 & 8 represent format of Route Request and Route Reply in AODV

The reason one can unicast R.REP back is that every node forwarding a R.REQ message caches a route back to the source node. Fig. 9 & 10 represent route establishment and descriptive table of AODV.
V. Working of Hybrid Routing Protocol

A. ZRP (Zone Routing Protocol)

Zone routing protocol is the hybrid routing protocol. It uses the advantages of both table-driven and on-demand routing protocol. ZRP aims to address excess bandwidth and long route request delay of proactive and reactive routing protocols. ZRP divides the entire network into variable size zones. Each zone associated with the node in the network. The size of a zone is not determined by geographical measurement but is given by a radius of length $\rho$, where $\rho$ is the number of hops to the perimeter of the zone.

i) Working of Zone Routing Protocol (ZRP)

A routing zone is defined for each node separately, and the zones of neighboring nodes overlap. The routing zone has a radius expressed in hops. The zone includes the distance from the node is at most hops [8].

An example of routing zone is shown in Figure 10, where the routing zone of S includes the nodes A–I, but not K. The radius is shown as a circle around the nodes. It should be noted that the zone is defined as hops, not as a physical distance.

The nodes of a zone are divided into peripheral nodes and interior nodes. Peripheral nodes are nodes whose minimum distance to the central node is exactly equal to the zone radius. The nodes whose minimum distance is less than are interior nodes. In Figure 10, the nodes A–F are interior nodes, the nodes G–J are peripheral nodes and the node K is outside the routing zone. Note that node H can be reached by two paths, one with length 2 and one with length 3 hops. The node is however within the zone, since the shortest path is less than or equal to the zone radius. ZRP uses proactive approach for routing inside the zone i.e. intra-zone routing protocol (IARP) and reactive approach for routing outside the zone i.e. inter-zone routing protocol (IERP).
ii) ZRP Architecture

![ZRP Architecture Diagram]

Fig11: ZRP Architecture

a) Intra Zone Routing Protocol (IARP): IARP is used by a node to establish communication with the internal nodes of its zone and is limited by the zone radius [11]. Routes are maintained inside the zone, each node continuously needs to update the routing information in order to determine the peripheral nodes as well as to maintain a map with which nodes can be reached locally.

b) Inter Zone Routing Protocol (IERP): IERP is used for the communication between nodes of different zones. It is reactive routing component of Zone Routing protocol which offers enhanced route discovery [11]. The IERP needs to be able to take advantage of the local connectivity provided by IARP. Route discovery is done through a process called Bordercasting that uses a Bordercast Routing Protocol (BRP) for transmission of route requests to peripheral nodes.

c) Bordercast Routing Protocol (BRP): BRP is used to direct the route requests initiated by the IERP to the peripheral nodes and also utilizes the topology information provided by IARP to construct a bordercast tree. For route requests away from areas of network, a query control mechanism is employed by BRP.

VI PERFORMANCE PARAMETERS

TABLE II: COMPARISON between Routing Protocols of MANET

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Reactive Protocol</th>
<th>Proactive Protocol</th>
<th>Hybrid Protocol</th>
</tr>
</thead>
<tbody>
<tr>
<td>Routing Scheme</td>
<td>On-Demand</td>
<td>Table-Driven</td>
<td>On-Demand/Table-driven</td>
</tr>
<tr>
<td>Routing Overhead</td>
<td>LOW</td>
<td>HIGH</td>
<td>High in comparison with Proactive/Proactive</td>
</tr>
<tr>
<td>Latency</td>
<td>High due to Flooding</td>
<td>Low due to routing table</td>
<td>Low latency due to traffic</td>
</tr>
<tr>
<td>Storage Capacity</td>
<td>Low generally depends upon number of routes</td>
<td>High due to routing tables, Low latency due to traffic</td>
<td>High</td>
</tr>
<tr>
<td>Mobility Support</td>
<td>Route Maintenance</td>
<td>Periodical Update</td>
<td>Both Route maintenance and periodical update</td>
</tr>
</tbody>
</table>

VII CONCLUSION

In this paper an attempt has been made to study and analyze three different routing protocols DSDV, DSR, AODV and ZRP. Generally on-demand protocols (DSR and AODV) perform better than DSDV. Especially when mobility increases. Even with lower mobility, DSDV suffer big packet loss. When it comes to power requirement proactive protocols has high power consumption than on-demand protocols. Whereas in case of Hybrid Routing Protocol it combines the advantages of both Reactive as well as Proactive Protocol. Inside the routing zone, proactive component IARP maintains the routing tables. Outside the zone, route discovery mechanism is done by reactive component IERP using route requests and route replies. A bordercasting process is used for route discovery. To reduce the amount of query traffic, query control mechanisms query detection and early termination can be used.

REFERENCES


INTERNET PROTOCOL VERSION 6(IPv6)

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Abstract.-IPv6 has become essential now days, as the networking world is spreading its area drastically. To understand IPv6 is being so important which could prepare us to meet today’s real world networking challenges. IPv6 is referred as the next generation Internet protocol and it is created to meet the solution of the problems that occurred in IPv4 module such as inevitable and impending address-exhaustion crises. IPv6 brings us flexibility, efficiency, capability, and optimized functionality for easier communication. This paper highlights the ground level of IPv6, solution of the problems that occurred in ex models and the future applications.

Keywords-- IPv4, IPv6, addressing format, addressing types

1. INTRODUCTION

IPv6 is the advanced internet protocol that was introduced in the networking system a decade before. It is the revision of all the internet protocols. In other words, Ipv6 is a new numbering system that provides more effective and efficient hierarchically communication system. It is a communication protocol which also provides with an identification and location system for computers on network and route traffic across the internet. Ipv6 helps to resolve the problems which were rising in the previous versions IP, i.e. IP address depletion and scaling of the routing.

2. REQUIREMENT OF IPv6 IN NETWORKING SYSTEM

Why IPv6 needed though we had IPv4 model? And the answers come from multiple directions. IPv4 has only about 4.3 billion addresses available till date and the requirement of IP addresses is increasing. There are about 7 billion people in the world today, and it’s estimated that only just over 10 per cent of that population is currently connected to the internet.

<table>
<thead>
<tr>
<th>S.N</th>
<th>Features</th>
<th>Internet protocol version 4(IPv4)</th>
<th>Internet protocol version 6(IPv6)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Deployed</td>
<td>1981</td>
<td>1999</td>
</tr>
<tr>
<td>2</td>
<td>Address Size</td>
<td>32-bit number</td>
<td>128-bit number</td>
</tr>
<tr>
<td>3</td>
<td>Prefix Notation</td>
<td>192.149.0.0/2</td>
<td>3FFE:F200:02</td>
</tr>
<tr>
<td>4</td>
<td>Number of Addresses</td>
<td>(\frac{2^{32}}{6} \approx 4,294,967,296)</td>
<td>(\frac{2^{128}}{34.48} \approx 340,282,366,920,938,463,463,374,607,431,768,211,456)</td>
</tr>
</tbody>
</table>

To communicate with the other devices with internet, an IP address is required to be assigned to every device. With the ever-increasing number of new devices being connected to the Internet, the need arose for more addresses than IPv4 is able to accommodate. IPv6 uses a 128-bit address, allowing \(2^{128}\) or approximately \(3.4 \times 10^{38}\) addresses, or more than \(7.9 \times 10^{28}\) times as many as IPv4, which uses 32-bit addresses. IPv4 allows only approximately 4.3 billion addresses. The two protocols are not designed to be interoperable, complicating the transition to IPv6.

IPv6 is the successor to the Internet’s first addressing infrastructure, Internet Protocol version 4 (IPv4). In contrast to IPv4, which defined an IP address as a 32-bit value, IPv6 addresses have a size of 128 bits. Therefore,
IPv6 has a vastly enlarged address space compared to IPv4.

3. IPv6 ADDRESSING, EXPRESSIONS AND PACKET FORMAT

A. Addressing
IPv6 addresses are represented as eight groups of four hexadecimal digits separated by colons, for example:
2001:0DB8:AC10:0E01:0000:0000:0000:0000

Fig 1[12] Representing the format of IPv6 address.

The 128 bits of an IPv6 address are represented in 8 groups of 16 bits each. Each group is written as 4 hexadecimal digits and the groups are separated by colons (:). The address 2001:0DB8:0000:0000:0000:0000:0000:0001 is an example of this representation.

Fig-2 IPv6 address sub parts

For convenience, an IPv6 address may be abbreviated to shorter notations by application of the following rules, where possible.

1) One or more leading zeroes from any groups of hexadecimal digits are removed; this is usually done to either all or none of the leading zeroes. For example, the group 0042 is converted to 42.

2) Consecutive sections of zeroes are replaced with a double colon (::). The double colon may only be used once in an address, as multiple uses would render the address indeterminate. RFC 5952 recommends that a double colon must not be used to denote an omitted single section of zeroes.

So the abbreviated address of the above example, according to the rules would be:
2001:db8::ff00:42:8329

The loopback address, 0000:0000:0000:0000:0000:0000:0000:0001, may be abbreviated to ::1 by using both rules.

B. Packet formatting
A IPv6 consist of a header and a payload. Packets require header in a fixed portion which have a minimal functionality. They can follow optional extension if special feature are required. The first 40 octets (i.e. 320 bits) of the IPv6 packet is occupied by the fixed header which contains the source and destination addresses, traffic classification options, a hop counter, and the type of the optional extension or payload which follows the header.

<table>
<thead>
<tr>
<th>Versions</th>
<th>Traffic class</th>
<th>Flow label</th>
</tr>
</thead>
<tbody>
<tr>
<td>Payload length</td>
<td>Next header</td>
<td>Hop limit</td>
</tr>
<tr>
<td>Source address</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Destination address</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Fig-3 Packet format of IPv6

There is an another field “i.e. Next Header” field which directs user to interpret the data which follows the header. The packet contains option, the next header contains the option type of the next option. The upper layer protocol that is carried in the public payload is pointed by the last option of the next header field.

The option that are carried by the extension header can be used for special treatment of a packet in the network. For example, it can be used fragmentation, routing, security using the IPsec framework.

The payload cannot be more than 64KB without using special options. Jumbo payload option exit in the payload must be within 4GB.
In IPv6, the routers never fragment a packet. Hosts are forced to make use of the path MTU discovery to make their packet small enough to reach the destination without fragmenting the packet.

C. Processing with device
In IPv6, the processing by routers is simplified i.e., the packet header and the process of packet forwarding have been simplified. The packet header in IPv6 is simpler than that used in IPv4, with many rarely used fields moved to separate optional header extensions. In IPv6 routers, fragmentation of the packet is not performed. IPv6 hosts are required to either perform path MTU discovery i.e., perform end-to-end fragmentation, or to send packets not larger than the IPv6 default MTU size of 1280 octets.

The IPv6 header is not protected by a checksum. Integrity protection is assumed to be assured by both link-layer and higher-layer (TCP, UDP, etc.) error detection. UDP/IPv4 may actually have a checksum of 0, indicating no checksum; IPv6 requires UDP to have its own checksum. Therefore, IPv6 routers do not need to recompute a checksum when header fields (such as Time To Live (TTL) or hop count) change. This improvement may have been made less necessary by the development of routers that perform checksum computation at link speed using dedicated hardware, but it is still relevant for software-based routers.

The TTL field of IPv4 has been renamed to Hop Limit in IPv6, reflecting the fact that routers are no longer expected to compute the time a packet has spent in a queue. In IPv6, the processing by routers is simplified i.e., the packet header and the process of packet forwarding have been simplified.

4. ADDRESS TYPES

A. Unicast
An IPv6 unicast address is an identifier for a single interface, on a single node. A packet that is sent to a unicast address is delivered to the interface identified by that address.

1) Aggregatable Global Address- An aggregatable global address is an IPv6 address from the aggregatable global unicast prefix. Aggregatable global IPv6 addresses are defined by a global routing prefix, a subnet ID, and an interface ID. The IPv6 global unicast address allocation uses the range of addresses that start with binary value 001 (2000::/3).

![Fig-4 Aggregatable Global Address Format.](image)

The aggregatable global address typically consists of a 48-bit global routing prefix and a 16-bit subnet ID or site-level aggregator (SLA).

2) Link-Local Address- A link-local address is an IPv6 unicast address that can be automatically configured on any interface using the link-local prefix FE80::/10 (1111 1110 10). IPv6 devices must not forward packets that have link-local source or destination addresses to other links.

![Fig-5 Link-Local Address Format](image)

3) Unique Local Address- A unique local address is an IPv6 unicast address that is globally unique and is intended for local communications. It is not expected to be routable on the global Internet and is routable inside of a limited area, such as a site. It may also be routed between a limited set of sites.

B. Anycast
An anycast address is an address that is assigned to a set of interfaces that typically belong to different nodes. A packet sent to an anycast address is delivered to the closest interface (as defined by the routing protocols in use) identified by the anycast address. Anycast...
addresses are syntactically indistinguishable from unicast addresses, because anycast addresses are allocated from the unicast address space. The nodes to which the anycast address is assigned must be explicitly configured to recognize that the address is an anycast address.

C. Multicast
A multicast address is also used by multiple hosts, which acquire the multicast address destination by participating in the multicast distribution protocol among the network routers. A packet that is sent to a multicast address is delivered to all interfaces that have joined the corresponding multicast group.

5. CONCLUSION
IPv6 is intended to replace IPv4, which still carries the vast majority of Internet traffic as of 2013. IPv6 will resolve all the problems and provide with a lot of applications. It deals with the long-anticipated problem of IPv4 address exhaustion. It also provides with the advantages to make the networking world more advance.
IPv6 gives more efficient routing and packet processing. IPv6 don’t support broadcast, therefore the bandwidth is consumed. Address auto-configuration (address assignment) is built in to IPv6 which also simplifies the network configuration.
Security level increases in IPv6 because IPSec (Internet Protocol Security), which provides confidentiality, authentication and data integrity, is baked into in IPv6. Because of their ability to carry malware, IPv4 ICMP packets are often blocked by corporate firewalls, but ICMPv6, the implementation of the Internet Control Message Protocol for IPv6, may be permitted as IPSec can be applied to the ICMPv6 packets. IPv6 has significant cost advantages in current networks, and in developing the larger scale networks required by industry and help to build a reliability communication system between the people more efficiently.

REFERENCES
Simulation Tools for Mobile Ad-hoc Network

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Abstract— Network simulation is the most useful and frequent methodology used to calculate different network topologies without real world accomplishment. Network simulators are commonly used in the research area to evaluate new theories and suggestions. There are a number of network simulators, for example, ns-2, ns-3, OMNET++, SWAN, OPNET, Jist, and GloMoSiM etc. Therefore, the selection of a network simulator for evaluating research work is a critical task for researchers. The major focus of this paper is to compare the open source network simulators based on the following parameters: CPU utilization, memory usage, computational time, and scalability by simulating a MANET routing protocol, to identify an optimal network simulator for the research community.

Keywords: Simulation, Network simulators, simulators comparison, MANET Routing Protocol.

1. INTRODUCTION

Wireless technology is advancing fast and new enhancements are expected on a regular basis. New and untested protocols cannot be launched in the computer network due to uncertainty of its successful outcome. Therefore, the new protocols are tested with logical simulation tools. After the simulation, those protocols are implemented in the real world, if it show promising result. Logical modeling results are not accurate in terms of consumed energy, memory, and processing power. This is the main disadvantage of the logical/analytical modeling. Although real world implementation provides realistic results, the physical implementation is a time consuming procedure and expensive, because it requires more number of hardware and human resources. Alternatively, simulation is affordable and provides good results in a cost effective way to calculate the performance of planned protocols.

In the early, research work on communication networks involved both experimentation and mathematical modeling to prove feasibility and to establish bounds on expected performance of computer networks. However, in the last decade, computer networks have gone through a rapid revolution and have become too complicated for mathematical analysis. Computer-based simulation plays an important role in the research work to help the researchers and network designers to recognize the behaviour and performance of the networks and its protocols. Computer simulation is often used to test the planned capacity of networks and to meet customer requirements. In addition, simulation is also used to explore a wide range of potential protocol designs through rapid evaluation and iteration [1]. However, different simulators require variable time, memory and computation power for evaluating proposed protocols/techniques.

This paper presents performance comparison of four network simulators, i.e. ns-2, ns3, OMNET++, and GloMoSiM. These simulators are open source and well known network simulators in the research area. The objective of this paper is to help the researchers in choosing the most suitable simulators for their work in terms of memory usage, computation time and CPU utilization.

2. RELATED WORK

Selecting the suitable network simulator is a critical task for researchers. There are researchers who have been testing different routing protocols [2] in different simulators with different network parameters [3] to evaluate the exact performance of network protocols.

[4] compares ns-2 with OMNeT++ and QualNet by using radio propagation models. The architecture of ns-2 and TOSSIM are compared in [5]. In [6] and [7], the authors present performance comparison of different network simulators such as Java Sim, ns-2, and SSFNET [8], specially focuses on simulators that are
designed for sensor net- Works [9] presents a qualitative comparison of ns-2 and OPNET. Furthermore[10] presents a performance comparison of network simulators that are specially designed for VANETs.

Figure 1. Overview of the basic modules in OMNeT++[11]

The main difference between this paper and previously published papers is that we compare the most popular and open source simulators. Moreover, we perform the comparison by selecting the latest versions of simulators such as OMNET++ v4.2 and ns-3 v3.10. In addition, we use a MANET routing protocol, i.e. AODV, to evaluate the performance of the network simulators.

3. SIMULATION TOOLS

This section introduces the selected network simulators, i.e. ns-2, ns-3, OMNET++ and GloMoSim. These simulators are selected because of their high reputation in the research community.

3.1 ns-2

Network Simulator-2 [12] is an open source and discrete event network simulator. It is used for the simulation of network protocols with different network topologies. It is used for wired as well as wireless networks. NS-2 was built using C++ and provides the simulation interface through OTcl, an object-oriented dialect of Tcl. The user describes a network topology by writing OTcl scripts, and after that the main NS program simulates the topology with particular parameters. For the graphical view of the network network animator (NAM) is used. This is the most familiar and broadly used network simulator for research work. NAM interface contains control features that allow users to forward, pause, stop and play the simulation. The interface of ns-2 is shown in Figure 2.

In ns-2, arbitrary network topologies can be clear that are composed of routers, links and shared media [13]. The physical activities of the network are processed and queued in form of events, in a scheduled order. Then these events are processed as per scheduled time that increases along with the processing of events. However, the simulation is not real time; it is considered essential. [14].

Ns is a discrete event simulator embattled at networking research provided by USC/ISI [NS2]. It models system as events, which the simulator has, list of. The process is complete such way: “take next one, run it, until done”. Each event happens in an direct of simulated time, but takes an arbitrary amount of real time. Simulator design is separating the “data” from the control: C++ for “data” (per packet processing, core of ns, fast to run, detailed, complete control); and OTcl for control (simulation scenario configurations, periodic action, manipulating existing C++ objects, fast to write and change).

“Ns” Components are Ns the simulator itself and Nam the network animator that permit to imagine ns output and that provides a GUI interface to create ns scripts.
3.2 ns-3
The ns-3 project [15] was initiated in mid 2006 and is still under serious development. Ns-3 is also an open source, discrete-event network simulator. Ns-3 is considered as an alternate of ns-2, not an extension [16]. Like ns-2, it does not have an OTcl API. It is written in C++ language and python. Ns-3.10 is the latest version of ns-3 that supports parallel simulation and has an improved feature set. In addition to that ns-3 network simulations can be implemented in C++, while some parts of the simulation can also be written using Python. Ns-3 interface is shown in Figure 3.

![Figure 3: ns-3 Simulator Interface](image)

Ns-3 supports both simulation and emulation using sockets. It also generates p-cap traces which can assist in debugging. To examine network traffic, to reduce trace files, some standard tools can be used like wire shark. Ns-3 provides a practical environment and its source code is well structured [17].

3.3 OMNET++
Since September 1997 OMNET++ [18] has been presented and presently has a great number of users. Unlike ns-2 and ns-3, OMNET++ is not only planned for network simulations. It is also used for modeling of multiprocessors, distributed hardware systems and performance evaluation of complex software systems. However, it is generally used for computer networks simulation. OMNET++ is a common discrete event, component-based (modular) open architecture simulation framework. The interface of OMNET++ is shown in Figure 4.

![Figure 4: OMNET++ Simulator Interface](image)

The inspiration behind the development of OMNET++ was to produce a powerful open-source discrete event simulation tool that can be used by academic, educational and research-oriented commercial institutions for the simulation of computer networks, distributed and parallel systems [19]. OMNET++ distributions are accessible for both UNIX and Windows-based systems. It was developed using component-oriented approach that promotes structured and reusable models. In addition, OMNET++ has widespread graphical user interface (GUI) and intelligence support [20].

3.4 GloMoSiM
GloMoSiM stands for Global Mobile Information System Simulator. It is used for large scale wireless networks. GloMoSiM uses parallel discrete-event simulation based on Parsec [21]. In addition, GloMoSiM uses the Parsec compiler to compile the simulation of protocols [22].

GloMoSiM is able to simulate a network which contains thousands of nodes and heterogeneous communication links, for instance multicast and asymmetric links. In addition, GloMoSiM supports direct satellite communication, multi-hop wireless
communication, and most of the usual Inter-net protocols. It is a library-based sequential and parallel simulator that is designed only for wireless networks [22]. GloMoSiM has a scalable simulation library that is based on the Parsec simulation environment [23].

It is developed as a set of library modules, each of which simulates a specific wireless communication protocol in the protocol stack [24].

4. SELECTING A ROUTING PROTOCOL

The ad hoc on demand distance vector (AODV) routing algorithm is used to compare the performance of the simulators. AODV is selected because of its pre-availability in the selected net-work simulators. AODV is a reactive routing protocol that establishes the route on demand (when the route is required by the source node) and maintains the route as long as required by the source node. It avoids the count to infinity problem by using sequential numbers on route up-dates. In addition, AODV maintains time-based states at each node and discards routing entries that are not recently used. AODV is comparatively fast in terms of topological network changes and updates only the nodes that are affected by topological changes.

4.1. AODV

In AODV, the network remains silent unless a connection is needed by any node in the network. When a connection is required, the source node broadcasts a connection request. Referring to Figure 5, node A wants to communicate with node H, therefore, node A broadcasts a route request (RREQ) message in the network.

Every node in the network sends the RREQ message to its neighbours and records the previous node from where the request was received. When the destination node H receives the RREQ message, it sends back a uni-cast route reply (RREP) message to the source node through the node that delivered the RREQ message. The middle nodes that receives the RREP message sends it to the next node with the smallest distance towards the source node as shown in Figure 6. The entries that are not used in the routing tables are recycled after a time. If a link fails, a routing error message is sent back to the transmitting node and the route finding process is repeated.

The advantage of AODV is that it generates no extra traffic for communication along existing links. In addition, distance vector routing is simple and does not require much memory or calculation. However, AODV requires more time to set up a connection, and the initial communication to establish a route is higher than some other protocols.

The main advantage of this protocol is that the routes are established on demand, and destination sequence numbers are used to find the newest route towards the destination. The disadvantage of this protocol is that intermediate nodes can lead to inconsistent routes if the source sequence number is very old. If the intermediate nodes do not have the latest destination sequence number, old entries may exist. In addition, multiple Route Reply packets may be created in response to a single Route Request packet that can lead to high control packet overhead. Moreover, AODV leads to unnecessary bandwidth consumption.
5. Simulation and Results
To estimate the performance of the simulators, we simulated AODV routing protocol on selected simulators and evaluated the performance.

5.1. Simulation Setup
Before the start of a simulation, we make up connection establishments between predetermined nodes, e.g. node A sends data to node B. As the communication starts, the source node starts transmitting at a regular interval of 0.2 seconds. During the simulation, the number of nodes was varied from 400 to 2000. In addition, each simulation was executed for 500 seconds in a simulation.

The simulations tools were executed on the Linux platform.

Table 1: Simulation Scenrio

<table>
<thead>
<tr>
<th>SIMULATION/SCENARIO</th>
<th>Simulation Time</th>
<th>X, Y Dimensions</th>
<th>Mobility Model</th>
<th>Packet size</th>
<th>Number of nodes</th>
<th>Routing protocol</th>
</tr>
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<tbody>
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<td>400-2000</td>
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<tr>
<td>OMNET++ version</td>
<td>4.2</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Figure 7: Number of nodes vs Memory usage

5.2. Results and Discussion
The performance comparison was done based on the following parameters: memory usage (MB), CPU utilization (percent), scalability, and computation time (seconds).

5.2.1. Memory Usage
We simulated the AODV protocol for 500 seconds while varying the number of nodes from 400 to 2000. As shown in the Figure 7, ns-2 uses the highest amount of memory while ns-3 uses the lowest amount of memory compared to OMNET++ and GloMoSim. As the number of nodes increases, there is a linear growth in memory consumption for all simulators with minor difference. ns-3 was found to be the most efficient in memory usage among selected simulators.

5.2.2. CPU Utilization
CPU utilization was measured while varying the number of nodes. For all simulators, there is little effect on CPU utilization as the number of nodes increases.

Figure 8 shows that the CPU utilization of ns-2 and ns-3 is almost similar (5% variation) and is much higher compared to GloMoSiM and OMNET++.

GloMoSiM and OMNET++ shows a CPU utilization of only up to 35% with very little difference between them. The behavior of ns-2 and ns-3 was analyzed based on CPU utilization in detail by executing different applications in parallel with simulation tool. The simulations usually take a long time to execute while researchers use other applications, waiting for the results. We found that when other applications are executed in parallel, the CPU utilization of ns-2 and ns-3 drops to about 50%, hence allowing other applications to execute in parallel.
5.2.3. Computation Time

The computation time was calculated by simulating AODV protocol for 500 seconds while increasing the number of nodes.

As illustrated in Figure 9, ns-2 has the highest computation time. In addition, ns-2's computation time increases rapidly with increasing number of nodes, which means ns-2 is not scalable. For large number of nodes, it may take a very long time compared to the other simulators. The computation time of the other simulators is quite low compared to ns-2. In terms of computation time and scalability, ns-3 appears to be the most efficient.

6. CONCLUSION

In this paper, we evaluate the performance of four network simulators with respect to different parameters. Based on the simulation results, we conclude that ns-3, OMNET++, and GloMoSim are capable of carrying out large scale network simulations. ns-3 has proven to be the fastest simulator among the selected simulators in terms of computation time.

REFERENCES


WSN Clustering Protocol

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Abstract—Clustering shows great impact on network life time. Clustering is process to group the sensors for the purpose of saving energy. Cluster heads will send the collected information to base station. Base station collects all data which is called data aggregation. This paper represents the study of major protocol LEACH. Cluster formation is represented by experimental results on MATLAB.

Keywords—Clustering; wireless sensor networks; mutual exclusive clustering.

INTRODUCTION

Sensors are Micro electromechanical Systems having tiny microprocessors and low power radio technologies which can observe the environment and react to changes in the physical phenomena of their surrounding environments. Sensors nodes are limited in power, computation capability and memory. They Communicate via RF Signal. These sensors networked together over a wireless medium, to provide result of their sensing functionality which is called wireless sensor Networks. In other words Wireless Sensor Networks (WSN’s) define a special class of ad hoc network composed of a large number of nodes with sensing capability [21]. Wireless sensor networks are having limitation regarding power resources and computational capacity. Author defined WSN as WSN is Network of Wireless Nodes for purpose of monitoring certain phenomena of interest where base station collects data from all the nodes, and analyzes this data to draw conclusions about the activity in the area of interest

A sensor network is a deployment of massive numbers of small, inexpensive, self powered devices that can sense, compute, and communicate with other devices for the purpose of gathering local information to make global decisions about a physical environment. Wireless Sensor Networks consists of individual nodes that are able to interact with their environment by sensing or controlling physical parameter. these nodes have to Collaborate in order to fulfill their tasks as usually.

A wireless sensor (figure 1.1) is a collection of small randomly dispersed devices that provide three essential functions having the ability to monitor physical and environmental conditions, often in real time, such as temperature, pressure, light and humidity. The ability to operate devices such as switches, motors or actuators that

Control those conditions provide efficient, reliable communications via a wireless network.

COMPONENTS OF WIRELESS SENSOR NETWORK[18]

Unlike their ancestor ad-hoc networks, WSNs are resource limited, they are deployed densely, they are prone to failures, the number of nodes in WSNs is several orders higher than that of ad hoc networks, WSN network topology is constantly changing, WSNs use broadcast communication mediums and finally sensor nodes don’t have a global identification tags [18].

• Sensor Field: A sensor field can be considered as the area in which the nodes are placed.

• Sensor Nodes: Sensors nodes are the heart of the network. They are in charge of collecting data and routing this information back to a sink.

• Sink: A sink is a sensor node with the specific task of receiving, processing and storing data from the other sensor nodes. They serve to reduce the total number of messages that need to be sent, hence reducing the overall energy requirements
of the network. Sinks are also known as data aggregation points.

- Task Manager: The task manager also known as base station is a centralized point of control within the network, which extracts information from the network and disseminates control information back into the network. It also serves as a gateway to other networks, a powerful data processing and storage centre and an access point for a human interface. The base station is either a laptop or a workstation.

![Fig. 1. Components of Wireless Sensor Network](image)

Basically, each sensor node comprises sensing, processing, transmission, mobilizer, position finding system, and power units (some of these components are optional like the mobilizer). Sensor nodes are usually scattered in a sensor field, which is an area where the sensor nodes are deployed. Sensor nodes coordinate among themselves to produce high-quality information about the physical environment.

Each sensor node bases its decisions on its mission, the information it currently has, and its knowledge of its computing, communication, and energy resources. Each of these scattered sensor nodes has the capability to collect and route data either to other sensors or back to an external base station(s). A base-station may be a fixed node or a mobile node capable of connecting the sensor network to an existing communications infrastructure or to the Internet where a user can have access to the reported data.

### CLUSTERING

Wireless sensor networks are networks of wireless nodes that are deployed over an area for the purpose of monitoring certain phenomena of interest. The nodes perform certain measurements, process the measured data and transmit the processed data to a base station over a wireless channel. The base station collects data from all the nodes, and analyzes this data to draw conclusions about the activity in the area of interest.

Base station collects all data which is called data aggregation. In order to support data aggregation through efficient network organization, nodes can be partitioned into a number of small groups called clusters. Each cluster has a coordinator, referred to as a cluster head, and a number of member nodes. Clustering results in a two-tier hierarchy in which cluster heads (CHs) form the higher tier while member nodes form the lower tier. The member nodes report their data to the respective CHs. The CHs aggregate the data and send them to the central base through other CHs. Because CHs often transmit data over longer distances, they lose more energy compared to member nodes. The network may be re-clustered periodically.

Networks are categorized as flat network and hierarchical network. In flat networks each sensor is expected to transmit gathered information to base station, which cause problem of more energy consumption. In hierarchical networks each sensor will not transmit sensed information to base station instead groups of sensors will transmit their information to their first representative. Clustering is more or less implementation of hierarchical network. It’s a grouping of sensors in a way so that each sensor need not to forward its information to its base station instead it will send to its cluster head after that cluster head will aggregate the collected information from its cluster members and transmit this aggregated data to base station. Clustering is technique that is beneficial for less energy consumption of transmission so it increases scalability and lifetime of the network[5].

- 78 -
Heterogeneous network contain the different powered sensors from the initial stage. In heterogeneous networks high powered sensors are pre decided to be the cluster head. Homogeneous networks in which power of each sensor is same at starting level it is required that cluster head should be changed periodically in contrast to heterogeneous. Cluster head rotation in homogeneous networks is done via various clustering algorithms.

In present scenario there are many clustering algorithms which are divided into two categories centralized and distributive on basis of cluster head selection. Centralized algorithms are those in which Cluster heads are chosen by base station. Base station will allocate which sensor will be the cluster head for cluster. The advantage of centralized is that there is no message passing head ack over the network. Only broadcasted message from base station will be sufficient for informing cluster head example of such algorithm is LEACH-C, SHORT etc. Distributed algorithms decide their cluster head via message passing in between sensors. They decide in between of them which sensor will be the cluster head and also keep rotating their role of being cluster head example of such algorithm is well known traditional LEACH, ERA, RRCH, and DHAC etc. Some clustering algorithm worked on the basis of network structure they use the parameter of residual energy for deciding next cluster head example of such algorithm is HEED.

Clustering has been shown to improve network lifetime, a primary metric for evaluating the performance of a sensor network. Although there is no unified definition of “network lifetime,” as this concept depends on the objective of an application, Common definitions include the time until the first/last node in the network depletes its energy and the time until a node is disconnected from the base station.

**CLUSTERING OBJECTIVES**

Clustering algorithms in the literature varies in their objectives. Often the clustering objective is set in order to facilitate meeting the applications requirements. For example if the application is sensitive to data latency, intra and inter-cluster connectivity and the length of the data routing paths are usually considered as criteria for CH selection and node grouping. The following discussion highlights popular objectives for network clustering.

- **Load balancing**: Even distribution of sensors among the clusters is usually an objective for setups where CHs perform data processing or significant intra-cluster management duties. Given the duties of CHs, it is intuitive to balance the load among them so that they can meet the expected performance goals. Load balancing is a more pressing issue in WSNs where CHs are picked from the available sensors. In such case, setting equal-sized clusters becomes crucial for extending the network lifetime since it prevents the exhaustion of the energy of a subset of CHs at high rate and prematurely making them dysfunctional. Even distribution of sensors can also leverage data delay. When CHs perform data aggregation, it is imperative to have similar number of node in the clusters so that the combined data report becomes ready almost at the same time for further processing at the base-station or at the next tier in the network.

- **Fault-tolerance**: In many applications, WSNs will be operational in harsh environments and thus nodes are usually exposed to increased risk of malfunction and physical damage. Tolerating the failure of CHs is usually necessary in such applications in order to avoid the loss of important sensors’ data. The most intuitive way to recover from a CH failure is to re-cluster the network. However, re-clustering is not only a resource burden on the nodes, it is often very disruptive to the on-going operation. Therefore, contemporary Fault-tolerance techniques would be more appropriate for that sake. Assigning backup CHs is the most notable scheme pursued in the literature for recovery from a CH failure. The selection of a backup and the role such spare CH will play during normal network operation varies. When CHs have long radio range, neighboring CHs can adapt the sensors in the failing cluster. Rotating the role of CHs among nodes in the cluster can also be a means for fault-tolerance in addition to their load balancing advantage.

- **Increased connectivity and reduced delay**: Unless CHs have very long-haul communication capabilities, e.g. a satellite link, inter-CH connectivity is an important requirement in many applications. This is particularly true when CHs...
are picked from the sensors population. The connectivity goal can be just limited to ensuring the availability of a path from every CH to the base-station or be more restrictive by imposing a bound on the length of the path. When some of the sensors assume the CH role, the connectivity objective makes network clustering one of the many variant of the connected dominating set problem. On the other hand, when data latency is a concern, intra-cluster connectivity becomes a design objective or constraint.

- Minimal cluster count: This objective is particularly common when CHs are specialized resource-rich nodes. The network designer often likes to employ the least number of these nodes since they tend to be more expensive and vulnerable than sensors. For example, if CHs are laptop computers, robots or a mobile vehicle there will be inherently some limitation on the number of nodes. The limitation can be due to the complexity of deploying these types of nodes, e.g. when the WSN is to operate in a combat zone or a forest. In addition, the size of these nodes tends to be significantly larger than sensors, which makes them easily detectable. Node visibility is highly undesirable in many WSNs applications such as border protection, military reconnaissance and infrastructure security.

- Maximal network longevity: Since sensor nodes are energy-constrained, the network’s lifetime is a major concern especially for applications of WSNs in harsh environments. When CHs are richer in resources than sensors, it is imperative to minimize the energy for intra-cluster communication. If possible, CHs should be placed close to most of the sensors in its clusters. On the other hand, when CHs are regular sensors, their lifetime can be extended by limiting their load as we mentioned earlier. Combined clustering and route setup has also been considered for maximizing network’s lifetime. Adaptive clustering is also a viable choice for achieving network longevity.

In this paper we in section II present some existing clustering protocols and their comparison following section III is proposal of new protocol. Section IV presents the experimental results and followed by the conclusion in section V.

LEACH

LEACH (Low Energy Adaptive Clustering Hierarchy) is a self-organizing, adaptive clustering-based protocol that uses randomized rotation of cluster-heads to evenly distribute the energy load among the sensor nodes in the network. LEACH collects data from distributed micro-sensors and transmits it to a base station. LEACH uses the following clustering-model. Some of the nodes elect themselves as cluster-heads. These cluster-heads collect sensor data from other nodes in the vicinity and transfer the aggregated data to the base station. Since data transfers to the base station dissipate much energy, the nodes take turns with the transmission the cluster-heads rotate [17]. This rotation of cluster-heads leads to a balanced energy consumption of all nodes and hence to a longer lifetime of the network [17].

LEACH works based on these basic assumptions

- Base station is fixed means immobile and located far away from the sensors
- All nodes in the network are homogeneous and energy constrained.
- All nodes are able to reach BS.
- Nodes have no location information.
- Channels are Symmetric propagation.
- Cluster-heads perform data compression.

LEACH protocol architecture[5,17]

LEACH (Low Energy Adaptive Clustering Hierarchy) is a distributed clustering protocol which utilizes randomized rotation of local CHs to evenly distribute energy utilization between the nodes of WSNs. The whole operation of the LEACH protocol is divided into rounds. Each round consists of Set-up phase (clusters are organized) in which Cluster Head Selection. Followed by Cluster Formation Steady state Phase (data transmission).

SET-UP PHASE

At the beginning of each round, each node advertises it probability, r (depending upon its current energy level) to be the Cluster Head, to all other nodes. Nodes (k for each round) with higher probabilities are chosen as Cluster Heads. Cluster Heads broadcasts an advertisement message (ADV) using CSMA MAC protocol. Based on the received signal strength, each non-Cluster Head node determines its Cluster Head.
for this round (random selection with obstacle). Each non-Cluster Head transmits a join-request message (Join-REQ) back to its chosen Cluster Head using a CSMA MAC protocol. Cluster Head node sets up a TDMA schedule for data transmission in the cluster.

**STEADY-STATE PHASE:**

TDMA schedule is used to send data from node to head cluster. Head Cluster aggregates the data received from node clusters. Communication is via direct-sequence spread spectrum (DSSS) and each cluster uses a unique spreading code to reduce inter-cluster interference. Data is sent from the cluster head nodes to the BS using a fixed spreading code and CSMA. The cluster-head node, after receiving all the data, aggregates it before sending it to the BS. After a certain time, which is determined a priori, the network goes back into the setup phase again and enters another round of selecting new CH.

LEACH is completely distributed and requires no global knowledge of network. The number of nodes that remain alive using LEACH is significantly larger (four to eight times larger) than that using static clustering or minimum transmission energy (MTE) routing.

To meet the unique requirements of wireless micro sensor networks, we developed LEACH, application-specific protocol architecture [5,17]. The application that typical micro sensor networks support is the monitoring of a remote environment. Since individual nodes’ data are often correlated in a micro sensor network, the end user does not require all the (redundant) data; rather, the end user needs a high-level function of the data that describes the events occurring in the environment. Because the correlation is strongest between data signals from nodes located close to each other, we chose to use a clustering infrastructure as the basis for LEACH. This allows all data from nodes within the cluster to be processed locally, reducing the data set that needs to be transmitted to the end user. In particular, data aggregation techniques can be used to combine several correlated data signals into a smaller set of information that maintains the effective data (i.e., the information content) of the original signals. Therefore, much less actual data needs to be transmitted from the cluster to the base station (BS).

For the development of LEACH, we made some assumptions about the sensor nodes and the underlying network model. For the sensor nodes, we assume that all nodes can transmit with enough power to reach the BS if needed, that the nodes can use power control to vary the amount of transmit power, and that each node has the computational power to support different MAC protocols and perform signal processing functions. These assumptions are reasonable due to technological advances in radio hardware and low-power computing. For the network, we use a model where nodes always have data to send to the end user and nodes located close to each other have correlated data.

Although LEACH is optimized for this situation, it will continue to work if it were not true. In LEACH, the nodes organize themselves into local clusters, with one node acting as the cluster head. All non-cluster head nodes transmit their data to the cluster head, while the cluster head node receives data from all the cluster members, performs signal processing functions on the data (e.g., data aggregation), and transmits data to the remote BS. Therefore, being a cluster head node is much more energy intensive than being a non-cluster head node. If the cluster
heads were chosen a priori and fixed throughout the system lifetime, these nodes would quickly use up their limited energy. Once the cluster head runs out of energy, it is no longer operational, and all the nodes that belong to the cluster lose communication ability. Thus, LEACH incorporates randomized rotation of the high-energy cluster head position among the sensors to avoid draining the battery of any one sensor in the network. In this way, the energy load of being a cluster head is evenly distributed among the nodes.

The operation of LEACH is divided into rounds. Each round begins with a set-up phase when the clusters are organized, followed by a steady-state phase when data are transferred from the nodes to the cluster head and on to the BS.

**EXPERIMENTAL RESULTS**

Performance of LEACH protocol has been measured over MATLAB with following Parameters. \( x_m=100, \ y_m=100, e_o=0.05, n=100, E_TX=50*0.000000000001, E_RX=50*0.00000000001, E_{fs}=10*0.000000000001, \) 
\( E_{DA}=5*0.000000000001, E_{adv}=50*0.000000000001, \) 
\( E_{RC}=15, \ E_{PS}=32. \) 
where \( x_m \) and \( y_m \) is used for field, \( e_o \) is initial energy of a sensors, \( n \) is number of sensors over random field, \( E_{TX} \) is transmission energy usage, \( E_{RX} \) is receiving energy usage, \( E_{fs} \) is free space communication energy usage, \( E_{DA} \) is data aggregation energy, \( E_{adv} \) is energy used for advertisement, \( E_{RC} \) is range of communication, \( E_{PS} \) is communication packet size and \( E_{adv} \) is size of advertisement packet.

**CONCLUSION**

Clustering plays vital role in network life. Network life is dependent on power of sensors. There are various clustering protocols in WSN some of which works probability base or on experience base and some are also working on behalf of residue energy of nodes or network. This paper presents cluster formation according to LEACH clustering protocol. Experimental results are shown according to LEACH working.
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- 83 -
Provide Security b/w Sender & Receiver in Bluetooth using Java

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Abstract :- Bluetooth has been developed to provide mobile ad hoc connectivity between a wide range of portable and fixed devices. Bluetooth is a Wireless Personal Area Network (WPAN) Standard that is moderately secure but still has weaknesses in its security architecture. The Bluetooth specification is ideal for mobile professionals who need to link notebook computers, mobile phones, PDA’s, PIMs, digital cameras, and other hand-held devices to do business at home, on the road, or in the office. Bluetooth provides three low power modes to conserve battery life: sniff mode, hold mode, and park mode. One such weakness has been identified in its pairing mechanism, which leads to an attacker guessing PIN number leading to the guessing of Initialization, Link & Encryption keys. Hence a need arises to make key generation mechanism robust against Pin guessing attack that is even if PIN Number is guessed; it is substantially difficult to crack Initialization, Link & Encryption Keys. This paper proposes a new robust approach for pairing and authentication mechanism, by using another shared secret parameter called ‘au_id’ (Authentication ID). It aims at low power consumption and provides security for both stationary and mobile devices. The purpose of this article is to give a good introduction that how to use java for the Bluetooth wireless technology, including an overview of its profiles. I’ll also cover the classes and methods of JSR-82, the official Java Bluetooth API. Finally, I’ll wrap things up by describing what software that I’ll need.

Keywords - Bluetooth Security, Authentication ID, PIN, java, java API, Key Generation

I. INTRODUCTION

In Bluetooth® technology is the global wireless standard enabling convenient, secure connectivity for an expanding range of devices and serves. It is an essential element for bringing everyday objects into the connected world.

Bluetooth is a proposed standard for wireless personal area networking and short range cable replacement. In Bluetooth, special effort has been taken to develop and standardize adequate security mechanisms and procedures for protecting the wireless radio link [2], [4], what the Bluetooth technology aims at – a cable replacement technology!

HISTORY

Bluetooth technology is the result of the joint achievements of nine leading companies: 3com, Lucent technologies, IBM, Intel, Microsoft, Motorola, Nokia, Toshiba, Ericsson altogether known as the Blue Tooth Special Interest Group (SIG), which has widespread participation by many companies. Originally initiated by L M Ericsson, it was designed as a short-range communications medium for wireless headsets to communicate with cellular phones. Bluetooth technology is the result of the joint achievements of nine leading companies: 3com, lucent technologies, IBM, Intel, Microsoft, Motorola, Nokia, Toshiba, Ericsson altogether known as the Blue Tooth Special Interest Group (SIG), which has widespread participation by many companies. Originally initiated by L M Ericsson, it was designed as a short-range communications medium for wireless headsets to communicate with cellular phones. Bluetooth technology is the result of the joint achievements of nine leading companies: 3com, lucent technologies, IBM, Intel, Microsoft, Motorola, Nokia, Toshiba, Ericsson altogether known as the Blue Tooth Special Interest Group (SIG), which has widespread participation by many companies.
Originally initiated by L M Ericsson, it was designed as a short-range communications medium for wireless headsets to communicate with cellular phones.

1.1 How does Bluetooth technology differ from other radio technologies?
Mobile phones, FM radio and television all use radio waves to send information wirelessly. And while Bluetooth technology also uses radio waves, it transmits them over a shorter distance.

Radios and TV broadcasts over many miles or kilometers. Bluetooth technology sends information within your Personal Area Network or "PAN" (aka your own personal space) at distances up to 100 meters (328 feet)—depending upon device implementation. Bluetooth technology operates in the unlicensed industrial, scientific and medical (ISM) band at 2.4 to 2.485 GHz, using a spread spectrum, frequency hopping, full-duplex signal at a nominal rate of 1600 hops/sec.

1.2 Traffic control by Bluetooth enable mobile phone:

It enables a mobile phone with Bluetooth to control a computer system and all the hardware attached to it. Bluetooth enabled remote control is a remote control for a personal computer. Your mobile phone acts as a remote control for the applications present in your computer. For instance, you can know the details about road map and emergency services present in your computer with the help of your mobile phone. Each Bluetooth hardware requires a program called Bluetooth stack to be installed before use. Different vendors have different Bluetooth stacks and most of the time they are incompatible with each other. We have tried to make Bluetooth enabled Remote Control generic and successfully achieved it.

2. JAVA FOR MOBILE DEVICES

Java ME technology was originally created in order to deal with the constraints associated with building applications for small devices. The basics for Java ME technology to fit such a limited environment and make it possible to create Java applications running on small devices with limited memory, display and power capacity.

Java ME platform is a collection of technologies and specifications that can be combined to construct a complete Java runtime environment specifically to fit the requirements of a particular device or market. This offers a flexibility and co-existence for all the players in the ecosystem to seamlessly cooperate to offer the most appealing experience for the end-user. The Java ME technology is based on three elements:

i. A configuration provides the most basic set of libraries and virtual machine capabilities for a broad range of devices,

ii. A profile is a set of APIs that support a narrower range of devices, and

iii. An optional package is a set of technology-specific APIs.

iv. Over time the Java ME platform has been divided into two base configurations, one to fit small mobile devices and one to be targeted towards more capable mobile devices like smart-
phones and set top boxes. The configuration for small devices is called the Connected Limited Device Configuration (CLDC) and the more capable configuration is called the Connected Device Configuration (CDC). The figure 2 below represents an overview of the components of Java ME technology and how it relates to the other Java Technologies.[6][7]

B. JAVA BLUETOOTH API

Bluetooth hardware has advanced, there has been no standardized way to develop Bluetooth applications – until JSR 82 came into play. It is the first open, non-proprietary standard for developing Bluetooth applications using the Java programming language. It hides the complexity of the Bluetooth protocol stack behind a set of Java APIs that allow you to focus on application development rather than the low-level details of Bluetooth. JSR 82 is based on version 1.1 of the Bluetooth Specification. Like all JSRs, the Java APIs for Bluetooth are being developed through the Java Community Process. JSR 82 consists of two optional packages: the core Bluetooth API and the Object Exchange (OBEX) API. The latter is transport-independent and can be used without the former.

Note: The Java APIs for Bluetooth do not implement the Bluetooth specification, but rather provide a set of APIs to access and control a Bluetooth-enabled device. JSR 82 concerns itself primarily with providing Bluetooth capabilities to J2ME-enabled devices.

The Java APIs for Bluetooth target devices with the following characteristics:

512K minimum of total memory available (ROM and RAM) (application memory requirements are additional) Bluetooth wireless network connection Compliant implementation of the J2ME Connected Limited Device Configuration (CLDC)

3. Security overview between two devices:

The security architecture presented in this paper provides a very flexible security framework. This framework dictates when to involve a user (e.g., to provide a PIN) and what actions the underlying BT protocol layers follow to support the desired security check-ups. Within this framework, a number of realizations of the presented architecture can be instantiated, some of them simpler and some of them more advanced than the one discussed in detail in this paper, without moving outside the scope of the architecture.

4. Bluetooth Sender and receiver code:

Here is some code to illustrate how to communicate between two NXT bricks via the build-in Bluetooth. The code follows the following protocol between a ‘sender’ and ‘receiver’ NXT:

2. Receiver: waits for connection
3. Sender: connect to receiver
4. Receiver: waits for data
5. Sender: sends an integer
6. Receiver: waits for data
7. Sender: sends an integer
8. Sender: waits for data
9. Receiver: sends an integer
10. Sender & Receiver: close connection

Fig 3. Security architecture

11. Receiver: waits for connection
12. **Sender**: connect to receiver
13. **Receiver**: waits for data
14. **Sender**: sends an integer
15. **Receiver**: waits for data
16. **Sender**: sends an integer
17. **Sender**: waits for data
18. **Receiver**: sends an integer
19. **Sender & Receiver**: close connection

Here is some code to illustrate how to communicate between two NXT bricks via the build-in Bluetooth. The code follows the following protocol between a ‘sender’ and ‘receiver’ NXT:

In an actually useful program you would put the code into methods like ‘send’ and ‘receive’ and store relevant data in state variables for further processing instead of just displaying it on the screen. To try the code ‘as is’, create two programs. Upload one to the ‘sender’, the other to the ‘receiver’, then follow the instructions.

* Test of NXT to NXT Bluetooth communication

* -------------------------------

* One NXT (the ‘sender’) connects to another NXT, sends 2 integers, and
* waits for a single integer as reply. Then it closes the connection.

* The other NXT (the ‘receiver’) waits for an NXT to establish a connection.
* If it receives a connection, it waits for two integers. After receiving
* two integers it sends a single integer back and closes the connection. For
* this to work:

* each NXT must have a unique name
* the ‘receiver’ must be a ‘known device’ for the ‘sender’ (see below)

* => the receiver must run BTReceive first and wait
* => the sender runs BTSend to get the chain of events going.

* To setup the 'sender' (before running the program):

* 1. Change the name string in BTSender.java to the name of your receiver NXT

* 2. Make sure the receiver is in the known devices list of the sender:

* a) turn on the receiver NXT
* b) make sure Bluetooth is on and the device is visible
* c) on the sender, select the Bluetooth menu and select Search
* d) the name of the receiver NXT should appear
* e) select Add to add it to the known devices of the sender

* You can check this has been done by selecting 'Devices' from the Bluetooth
* menu on the sender.

* 3. Compile, upload, and run 'BTReceive' on the receiver NXT

* 4. Specify correct name in sender code, then compile, upload, and run the
*    'BTSender' program on the sender NXT

*/

4.1 BTReceive Program
See above comments that describe the two classes working in conjunction with each other (perhaps copy the comments into these classes). Note that you must create two separate classes, one for program. Also, the last line to sleep for 4 seconds is only there so you can read the screen!

```java
import java.io.*;
import lejos.nxt.LCD;
import lejos.nxt.comm.BTConnection;
import lejos.nxt.comm.Bluetooth;
public class BTReceive
{
public static void main(String[] args) throws Exception
{
    LCD.clear();
    LCD.drawString("Receiver wait...", 0, 0);
    LCD.refresh();
    try
    {
        BTConnection connection = Bluetooth.waitForConnection();
        if (connection == null)
            throw new IOException("Connect fail");
        LCD.drawString("Connected.", 1, 0);
        DataInputStream input = connection.openDataInputStream();
        DataOutputStream output = connection.openDataOutputStream();
        int answer1 = input.readInt();
        LCD.drawString("1st = " + answer1, 2, 0);
        int answer2 = input.readInt();
        LCD.drawString("2nd = " + answer2, 3, 0);
        output.writeInt(0);
        output.flush();
        LCD.drawString("Sent data", 4, 0);
        input.close();
        output.close();
        connection.close();
        LCD.drawString("Bye ...", 5, 0);
    }
    catch(Exception ioe)
    {
        LCD.clear();
        LCD.drawString("ERROR", 0, 0);
        LCD.drawString(ioe.getMessage(), 2, 0);
        LCD.refresh();
    }
    Thread.sleep(4000);
}
}
```

4.2 BTSend Program

```java
import java.io.*;
import javax.bluetooth.RemoteDevice;
import lejos.nxt.LCD;
import lejos.nxt.comm.BTConnection;
import lejos.nxt.comm.Bluetooth;
public class BTSend
{
public static void main(String[] args) throws Exception
{
    // Change this to the name of your receiver
    String name = "MyNXT";
    LCD.clear();
```
LCD.drawString("Connecting...", 0, 0);
LCD.refresh();
try
{
    RemoteDevice receiver = Bluetooth.getKnownDevice(name);
    if (receiver != null)
        throw new IOException("no such device");
    BTConnection connection = Bluetooth.connect(receiver);
    if (connection == null)
        throw new IOException("Connect fail");
    LCD.drawString("connected.", 1, 0);
    DataInputStream input = connection.openDataInputStream();
    DataOutputStream output = connection.openDataOutputStream();
    output.writeInt(42);
    output.writeInt(-42);
    output.flush();
    LCD.drawString("Sent data", 2, 0);
    LCD.drawString("Waiting...", 3, 0);
    int answer = input.readInt();
    LCD.drawString("# = " + answer, 4, 0);
    input.close();
    output.close();
    connection.close();
    LCD.drawString("Bye...", 5, 0);
}

} catch(Exception ioe)
{
    LCD.clear();
    LCD.drawString("ERROR", 0, 0);
    LCD.drawString(ioe.getMessage(), 2, 0);
    LCD.refresh();
    Thread.sleep(4000);
}

5. CONCLUSION

This article presented a tutorial on the Java for Bluetooth wireless technology. The sample code demonstrated how easy it is to develop wireless applications for Bluetooth-enabled devices. After all of the preliminaries are out of the way, i can stream data back and forth to any Bluetooth-enabled device in our area, whether it's running Java or not.

6. REFERENCES

Review of Routing Protocol in MANET’s

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Abstract : Mobile Ad-hoc networks [MANET] has been generally regarded as an ideal network model for group communication, because of its speciality of instant establishment. However the security of MANET is still challenging issue. Although there are some existing security schemes such as ARAN (Authenticated routing for Ad-hoc networks).Protocol that makes use of cryptographic certificate to provide end–to-end authentication during the routing phases, the overhead of security computation is still a serious hurdle for real application. This paper proposes a security mechanism for the Ad-hoc On-demand Distance Vector(AODV) routing protocol. The three main techniques including digital signature , One way hash Function and Double one way hash function verification are introduced to ensure the authentication . Nonrepudiation and integrity of the important routing information is AODV protocol. The comparission with some existing secure AODV protocol demonstrates that expends the security scope and guards against several attacks of their ranges.

Keywords:- Ad-hoc Networking_ Distance Vector Routing_ Dynamic Routing_ Mobile Networking_ Wireless Networks.

1. INTRODUCTION

With the staggering growth of wireless handheld devices and plummeting costs of mobile tele-communications, mobile ad hoc network has emerged as a major area of research for both the academic and the industrial sectors. A mobile ad-hoc network (MANET) is built on the fly where a number of mobile nodes work in cooperation without the engagement of any centralized access point or any fixed infrastructure. MANETs are self-organizing, self-configuring, and dynamic topology networks, which form a particular class of multi-hop networks. Minimal configuration, absence of infrastructure, and quick deployment make them convenient for combat, medical, and other emergency situations. All nodes in a MANET are capable of movement and can be interconnected in an arbitrary manner.

The issue of routing in MANET is somewhat challenging and non-trivial. Basically, the major challenges for routing in MANET are imposed by the resource constraints and mobility of the nodes participating in the network. As there is no fixed infrastructure in such a network, we consider each node as a host and a router at the same time. Hence, during routing of data packets within the network, at each hop, each host also has to perform the tasks of a router. In fact, these special aspects of mobile ad hoc networks have attracted many researchers to work on solving the routing issues in MANET.

So far, a significant number of proposals for routing in MANET have seen the daylight. However, We will know various types of routing schemes those are already proposed or those could be applied for these types of networks. Considering the practical scenarios, we will also discuss how the reality might betray the expectations.

On the other hand, the wide-open environment makes this network super vulnerable to inside and outside Attacks Especially in the case of routing [4], since the absence of central control, it is extremely difficult to prevent nodes from behaving improperly. Although there exist a large number of MANET routing protocols [3,4 ,5, 8,11], most of them were designed without any security considerations (generally it is assumed that all nodes are friendly). Besides, the resource constraints (both computation and bandwidth) of MANET put up great difficulties over the deployment of security. Two widely known reactive routing protocols are AODV (Ad-hoc On-Demand Distance Vector Routing) [4] and DSR(Dynamic Source
Routing) [4], which are both very efficient but are subject to a variety of attacks.

To reinforce the security of routing, ARAN makes use of cryptographic techniques to offer security in an open-manage environment. Since the security is based on public key cryptography, the efficiency of ARAN is under suspicion. In this paper, we pursue the advantages of one-time signature, which is more efficient in signing and verification, to replace conventional digital signature in protecting routing packets, though, at the same time, maintaining the same level authentication.

2. ROUTING PROTOCOL
In this section, we will learn about various routing protocols for MANET, their major aspects, and their relative pros and cons.

Ad-Hoc network is called as Mobile Ad-Hoc Network (MANET) because of mobility of nodes in network. They are IBSS (Independent Basic Service Set), because they does not need AP(Access Point) for communication in nodes. MANETs is a self-configuring network and form an uninformed topology. These nodes behave like routers in network to route the packet. MANETs are used in those areas where wire and wireless infrastructures are unreachable. Due to rapid change of topology in MANETs, MANETs routing protocols are required. The routing protocol is required whenever the source needs to communicates with destination. MANETs routing protocols are classified as:-

A. Reactive protocols
B. Proactive protocols
C. Hybrid protocols

![Fig. 1 MANETs Routing Protocols][1]

One of the most interesting aspects for routing in MANET, which many research works have tried to solve is, whether or not the nodes in the network should keep track of routes to all possible destination, or instead keep track of only those destinations of immediate interest. Generally, a node in MANET does not need a route to a destination until the node is necessarily be the recipient of packets, either as the final destination or as an intermediate node along the path from the source to the destination. As this is still a controversial issue, we can assume that the mechanism should not be fixed for all types of settings, instead based on the situation and application at hand, any of the methods could be chosen.

A. Reactive Protocols:
Reactive protocols are also called as on demand driven reactive protocols. It is mainly used to find the route between source and destination as needed. As per the demand of source this routing protocol initiate route discovery, to find the route to the destination. Then this route is used for further communication [1], [2] e.g. AODV[11].

B. Proactive Protocols:
Proactive protocols also called as Table driven routing protocols. Each node maintains routing tables which are consistent and up-to-date containing routing information for every node in the network. Whenever new node is entered in the network or removes from the network, control messages are sends to neighboring nodes then they update their routing tables. This routing protocol uses link-state routing algorithms which frequently flood the link information about its neighbors. Proactive routing protocols are OSPF and OLSR [11][2].

C. Hybrid Routing Protocol:
Hybrid routing protocol have advantages of both proactive and reactive routing protocols. Firstly it behave like proactive routing protocol, because in starting nodes have tables. Then whenever nodes finds that they does not have route to destination, they start route discovery and behave like reactive routing protocols. Hybrid protocols are TORA and ZRP[11][2].
2.1 Overview of AODV”

Ad-hoc routing protocols are mainly categorized into two groups proactive and reactive routing protocol. Where as third protocol is derived from both of these and is known as hybrid routing protocol. Proactive protocols are the one which maintain up-to-date routing information about the network, they are also known as table driven protocol. These protocols provide good reliability but are not suitable for nodes moving with higher speed. Reactive protocols are generally known as on-demand routing protocols. They discover route on demand when packet is to be sent. Ad-hoc on demand vector routing protocol is one of the reactive protocol[10][6].

AODV uses broadcast route discovery mechanism. It requests a route when needed and does not require nodes to maintain routes to destination that are not actively used in communication. It relies on dynamically establishing route table entries at intermediate nodes. To maintain the most recent routing information between the nodes, it uses the concept of destination sequence number. AODV protocol works in two steps
* Path Discovery
* Path Maintenance

A) Route discovery:

When a source node desires to send a message to some destination node, and doesn’t have a valid route to the destination, it initiates a path discovery process to locate the other node. It broadcasts a route request (RREQ) control packet to its neighbors, which then forward the request to their neighbors, and so on, either the destination or an intermediate node with a new route to the destination is located. The AODV protocol utilizes destination sequence numbers to ensure that all routes contain the most recent route information. Each node maintains its own sequence number. During the forwarding process the RREQ intermediate nodes record the address of the neighbour from which the first copy of the broadcast packet is received in their route tables, thereby establishing a reverse path. Once the RREQ reaches the destination or an intermediate node with a fresh enough route, the destination or the intermediate node responds by unicasting a route reply (RREP) control packet back to the neighbour from which first received the RREQ[9][8]

B) Route Maintenance

A route discovered between a source node and destination node is maintained as long as needed by the source node. The destination node or some intermediate node moves, the node upstream of the break initiates Route Error (RERR) message to the affected active upstream neighbors/nodes. Consequently, these nodes propagate the RERR to their predecessor nodes. This process continues until the source node is reached. When RERR is received by the source node, it can either stop sending the data or reinitiate the route discovery mechanism by sending a new RREQ message if the route is still required[9],[7],[8].

2.2 WORKING OF AODV:

Route Requests (RREQs), Route Replies (RREPs), and Route Errors (RERRs) are the message types defined by AODV. These message types are received via UDP, and normal IP header processing applies. So, for instance, the requesting node is expected to use its IP address as the Originator IP address for the messages. For broadcast messages, the IP limited broadcast address (255.255.255.255) is used. This means that such messages are not blindly forwarded. However, AODV operation does require certain messages (e.g., RREQ) to be disseminated widely, perhaps throughout the ad hoc network. The range of dissemination of such RREQs is indicated by the TTL in the IP header. Fragmentation is typically not required. As long as the endpoints of a communication connection have valid routes to each other, AODV does not play any role. When a route to a new destination is needed, the node broadcasts a RREQ to find a route to the destination. A route can be determined when the
RREQ reaches either the destination itself, or an intermediate node with a ‘fresh enough’ route to the destination. A ‘fresh enough’ route is a valid route entry for the destination whose associated sequence number is at least as great as that contained in the RREQ. The route is made available by uncasting a RREP back to the origination of the RREQ. Each node receiving the request caches a route back to the originator of the request, so that the RREP can be unicast from the destination along a path to that originator, or likewise from any intermediate node that is able to satisfy the request. Nodes monitor the link status of next hops in active routes. When a link break in an active route is detected, a RERR message is used to notify other nodes that the loss of that link has occurred. The RERR message indicates those destinations (possibly subnets) which are no longer reachable by way of the broken link. In order to enable this reporting mechanism, each node keeps a "precursor list", containing the IP address for each its neighbors that are likely to use it as a next hop towards each destination. The information in the precursor lists is most easily acquired during the processing for generation of a RREP message, which by definition has to be sent to a node in a precursor list. If the RREP has a nonzero prefix length, then the originator of the RREQ which solicited the RREP information is included among the precursors for the subnet route (not specifically for the particular destination). A RREQ may also be received for a multicast IP address. In this document, full processing for such messages is not specified. For example, the originator of such a RREQ for a multicast IP address may have to follow special rules. However, it is important to enable correct multicast operation by intermediate nodes that are not enabled as originating or destination nodes for IP multicast address, and likewise are not equipped for any special multicast protocol processing. For such multicast-unaware nodes, processing for a multicast IP address as a destination IP address MUST be carried out in the same way as for any other destination IP address [4] and [1]. AODV is a routing protocol, and it deals with route table management. Route table information must be kept even for short-lived routes, such as are created to temporarily store reverse paths towards nodes originating RREQs. AODV uses the following fields with each route table entry:

- Destination IP Address
- Destination Sequence Number
- Valid Destination Sequence Number flag
- Other state and routing flags (e.g., valid, invalid, repairable, being repaired)
- Network Interface
- Hop Count (number of hops needed to reach destination)
- Next Hop
- List of Precursors
- Lifetime (expiration or deletion time of the route)

Managing the sequence number is crucial to avoiding routing loops, even when links break and a node is no longer reachable to supply its own information about its sequence number. A destination becomes unreachable when a link breaks or is deactivated. When these conditions occur, the route is invalidated by operations involving the sequence number and marking the route table entry state as invalid [3] and [8].

One distinguishing feature of AODV is its use of a destination sequence number for each route entry. The destination sequence number is created by the destination to be included along with any route information it sends to requesting nodes. Using destination sequence numbers ensures loop freedom and is simple to program. Given the choice between two routes to a destination, a requesting node is required to select the one with the greatest sequence number.

3. RELATED WORK

Trust is an important aspect of mobile ad-hoc networks (MANETs). It enables entities to cope with uncertainty and uncontrollability caused by the free will of others. Trust computations and management are highly challenging issues in MANETs due to computational complexity constraints, and the independent movement of component nodes. This prevents the direct application of techniques suited for other networks. In MANETs, an untrustworthy node can wreak considerable damage and adversely affect the quality and reliability of data.
Therefore, analyzing the trust level of a node has a positive influence on the confidence with which an entity conducts transactions with that node. In this work we present a detailed survey on various trust computing approaches that are geared towards MANETs. We highlight the summary and comparisons of trust based AODV in MANET approaches.

**Secure Ad-hoc on demand distance vector Routing (SAODV):**

A secure version of AODV called Secure AODV (SAODV). It provides features such as integrity, authentication, and non-repudiation of routing data. It incorporates two schemes for securing AODV. To preserve the collaboration mechanism of AODV, SAODV includes a kind of delegation feature that allows intermediate nodes to reply to RREQ messages. This is called the double signature: when a node A generates a RREQ message, in addition to the regular signature, it can include a second signature, which is computed on a fictitious RREP message towards A itself. Intermediate nodes can store this second signature in their routing table, along with other routing information related to node A. If one of these nodes then receives a RREQ towards node A, it can reply on behalf of A with a RREP message, similarly to what happens with regular AODV. To do so, the intermediate node generates the RREP message, includes the signature of node A that it previously cached, and signs the message with its own private key[2].

SAODV does not require additional messages with respect to AODV. Nevertheless, SAODV messages are significantly bigger, mostly because of digital signatures. Moreover, SAODV requires heavyweight asymmetric cryptographic operations: every time a node generates a routing message, it must generate a signature, and every time it receives a routing message (also as an intermediate node), it must verify a signature. This gets worse when the double signature mechanism is used, because this may require the generation or verification of two signatures for a single message. In the SAODV operations, SAODV allows to authenticate the AODV routing data. Two mechanisms are used to achieve this: hash chains and signatures [2] and [4] and [12].

**Trusted Ad-hoc On-demand distance vector Routing (TAODV):**

TAODV is secure routing protocol which uses cryptography technologies recommended to take effect before nodes in the establish trust relationships among one another[1]. The main salient feature of TAODV is that using trust relationships among nodes, there is no need for a node to request and verify certificates all the time. TAODV (Trusted AODV) has several silent features:

1. Nodes perform trusted routing behaviors mainly according to the trust relationships among them;
2. A node that performs malicious behaviors will eventually be detected and denied to the whole network.
3. The performance of the System is improved by avoiding requesting and verifying certificates at every routing step[1].

That protocol greatly reduces the computation overheads. Assume that the keys and certificates needed by these cryptographic technologies have been obtained through some key management procedures before the node performs routing behaviors. Some extra new fields are added into a node’s routing table to store its opinion about other nodes’ trustworthiness and to record the positive and negative evidences when it performs routing with others. The main advantages of embedding trust model into the routing layer of MANET, save the consuming time without the trouble of maintaining expire time, valid state, etc. which is important in the situation of high node mobility and invalidity. Trusted AODV are mainly three modules in the whole TAODV system: basic AODV routing protocol, trust model, and trusted AODV routing protocol. Based on trust model, the TAOD routing protocol contains such procedures as trust recommendation, trust combination, trust judging, cryptographic routing behaviors, trusted routing behaviors, and trust updating [1] and [6] and [9].

**3.1 Security Aware Ad hoc Routing (SAR):**

SAR protocol integrates the trust level of a node and the security attributes of a route to provide the integrated security metric for the requested route. [2]A Quality of Protection
(QoP) vector used is a combination of security level and available cryptographic techniques. It uses the timestamps and sequence numbers to stop the replay attacks. Interception and subversion threats can be prevented by trust level key authentication. Attacks like modification and fabrication can be stopped by verifying the digital signatures of the transmitted packet. The main drawbacks of using SAR are that it required excessive encrypting and decrypting at each hop during the path discovery. [2]The discovered route may not be the shortest route in the terms of hop-count, but it is secure.

Reliable Ad-hoc On-demand Distance Vector Routing (RAODV):

The existing AODV has been extended to RAODV by adding two types of control packets: Reliable Route Discovery Unit (RRDU) and RRDU Reply (RRDU_REP). The RRDU messages are control packets sent by the source node along with RRDU-ID, to the destination at regular intervals and RRDU_REP message is the response of RRDU by the destination to the source node. RRDU_REP can only be generated by the destination. There is no impersonation i.e. no node other than the destination, can generate RRDU_REP on behalf of the destination[2].

Reliability List (RL) field is also adding in the routing table entry. An entry in the RL has Source address, a field called Forward Data Packet Count (FDPC) and RRDU-ID, i.e. the triplet (Source address, FDPC, RRDU-ID). The Routing Table entry format of RAODV is same as that of AODV [2] except for the additional RL field. RAODV uses RREQ, RREP messages for route discovery and RERR, HELLO messages for route maintenance which is similar in AODV. In addition, RAODV also uses RRDU and RRDU_REP to help discover the path and for reliability maintenance. In RAODV the path discovery can be thought of as consisting of two phases. The phase I is same as AODV. Whenever a node wishes to communicate with another node it looks for a route in its table. If a valid entry is found for the destination it uses that path else the node broadcasts the RREQ to its neighbors to locate the destination[2].

CONCLUSION

In this paper, a set of novel security mechanisms based on the Ad-hoc On-Demand Distance Vector Routing (AODV) are proposed. Three techniques, including digital signature, one-way hash function and double one-way hash verification ensure the authentication, non-repudiation and integrity of important routing information in RREQ, RREP and RERR packet of AODV. When comparing to some current secure AODV protocols like ARAN, SAODV and SRAODV, expands the security scope of them and provides more security service, such as protection towards big sequence number flood attack and selfish increment of hop_count.

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A Comprehensive Survey on WiMAX Scheduling Approaches

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Abstract: The objective of the broadband wireless technologies is to ensure the end to end Quality of Service (QoS) for service classes. Broadband wireless access technology has unique advantages compared to wired technology like easy configuration, high scalability, lower maintenance cost, lower investment and also end-user preference. But the growth in the industry demanded a better and outstanding service that could satisfy user needs. Therefore Quality of Service plays a major role in differentiating from different technologies. Wimax, IEEE 802.16 based wireless scheme, not only concentrates on lowering the cost of wired connections by enhancing features but also focusing highly on QoS(Quality of Service) requirements. The performance of any network essentially depends on quality of service required and also on the scheduling scheme. In this paper, in Section I, we focus on analyzing essential QoS parameters. Further we give a brief of various scheduling approaches and finally a comparative study of these algorithms.

INTRODUCTION TO WIMAX ARCHITECTURE AND SCHEDULING

The much anticipated technology Wimax stands for worldwide interoperability for microwave access. It aims to provide business and consumer wireless broadband services on the scale of the metropolitan area network (MAN). It is based on Institute of Electrical and Electronics Engineers 802.16 standard [1], [2].

On the other hand QoS (Quality of Service) refers to a broad collection of networking technologies and techniques. The goal of QoS is to provide guarantees on the ability of a network to deliver predictable results. Elements of network performance within the scope of QoS often include availability (uptime), bandwidth (throughput), latency (delay), and error rate. WiMAX QoS is still an open field of research and development for both constructors and academic researchers. The standard should also maintain connections for users and guarantee a certain level of QoS. Scheduling is the key model in computer multiprocessing operating system. It is the way in which processes are designed priorities in a queue and provide mechanism for bandwidth allocation and multiplexing at the packet level. [15]

IEEE 802.16 defines two types of operating modes: PMP mode and mesh mode. In the PMP mode SSs are geographically scattered around the BS. The performance of IEEE 802.16 in the PMP mode is verified in[4][7]. Our system model is based on a time-division-duplex (TDD) mode. The frame of IEEE 802.16 is divided into the downlink subframe and the uplink subframe. The downlink subframe starts with preamble followed by frame control header (FCH), downlink map (DL-MAP), uplink map (UL-MAP) messages and downlink burst data. The frame structure is illustrated in Figure 1. The DLMAP message defines the start time, location, size and encoding type of the downlink burst data which will be transmitted to the SSs. Since the BS broadcasts the DLMAP message, every SS located within the service area decodes the DL-MAP information elements (IEs) indicating the data burst directed to that SS in the downlink sub frame. After the transmit/receive transition gap (TTG), the uplink sub frame follows the downlink sub frame. IEEE 802.16 provides many advanced features like adaptive modulation coding (AMC), frame fragmentation and frame packing.

In IEEE 802.16 the BS( Base Station) centrally allocates the channels in different slots to different SSs( Subscriber Stations) for uplink and downlink which in turn allocates these resources to the various connections they are supporting at that time. Since BS is aware of the channel state of sub channels for all SSs and therefore can exploit channel user diversity by
allocating different sub channels to different SSs as shown in the Fig1 as given below [4].

IEEE 802.16e is expected to provide QoS for fixed and mobile users. QoS depends upon a number of implementation details like scheduling, buffer management and traffic shaping. The responsibility of scheduling and BW management is to allocate the resources efficiently based on the QoS requirement of the service classes. There are five service classes which are defined in IEEE802.16e standard. They are as follows:

- Unsolicited Grant Services (UGS): Designed to support Constant bit rate services like voice applications
- Real Time Data Polling Services (RTPS): Designed to support real time services that generates variable size data packets on a periodic basis like MPEG but insensitive to delay
- Extended Real Time Polling Services (ERTPS): Designed to support real time applications with variable data rates which require guaranteed data and delay. Example: Voice Over Internet Protocol (VOIP) with silence suppression
- Non Real Time Polling Services (NRTPS): Designed to support non real time and delay tolerant services that require variable size data grant burst types on a regular basis such as File Transfer Protocol (FTP)
- Best Effort (BE): Designed to support data streams that do not require any guarantee in QoS such as Hyper Text Transfer Protocol (HTTP)[15] [8].

<table>
<thead>
<tr>
<th>QoS Category</th>
<th>Applications</th>
<th>QoS Specifications</th>
</tr>
</thead>
<tbody>
<tr>
<td>UGS Unsolicited Grant Service</td>
<td>VoIP</td>
<td>-Maximum Sustained Rate -Maximum Latency Tolerance -Time Tolerance</td>
</tr>
<tr>
<td>RTPS</td>
<td>Streaming Audio or Video</td>
<td>-Maximum Sustained Rate -Maximum Latency Tolerance -Time Tolerance</td>
</tr>
<tr>
<td>ERTPS</td>
<td>Voice with Activity Detection (VoIP)</td>
<td>-Minimum Reserved Rate -Maximum Sustained Rate -Maximum Latency Tolerance -Time Tolerance</td>
</tr>
<tr>
<td>NRTPS</td>
<td>File Transfer Protocol (FTP)</td>
<td>-Minimum Reserved Rate -Maximum Sustained Rate -Maximum Latency Tolerance -Time Tolerance</td>
</tr>
<tr>
<td>BE</td>
<td>Data Transfer, Browsing, Web etc.</td>
<td>-Minimum Sustained Rate -Maximum Latency Tolerance -Time Tolerance</td>
</tr>
</tbody>
</table>

Table 1: WiMAX applications and QoS specifications [9]

The rest of the study is organized in the following way. Section 2 is the survey about related existing work. Section 3 is the brief about all scheduling algorithms and Section 4 gives a final comparison of various scheduling schemes.

**LITERATURE REVIEW**

Kim and Lim proposed a new scheduling scheme reflecting the delay requirement. Specifically, the authors added the delay...
requirement term in the proportional fair scheduling scheme and the scheduling parameters are optimised with respect to the QoS requirement. Therefore the QoS requirement is achieved without the excessive resource consumption. Vikram Mehta, Dr. Neena Gupta paper focuses on analyzing essential QoS parameters for WiMax Network. Their results help in critically analyzing QoS parameters for WiMax Network and it has been found that an optimum value of QoS parameters is obtained with increasing number of mobile nodes for WiMax Network.

Raina Triveni et. al. proposed a new downlink scheduling scheme reflecting the delay requirement of rtPS connections with respect to the various nrtPS and BE connections to achieve the optimal QoS requirement. Borin and Fonseca (2009) proposed a standard compliant scheduling solution for uplink traffic in IEEE 802.16 networks but wireless channel characteristics are not considered in this solution. Different scheduling algorithms have been compared in (Arhaif, 2011) and evaluated using Qualnet 5.0. The Different enabled (Diffserv), Round Robin (RR), Self Cloaked Fair (SCF), Strict Priority (SP), Weighted Round Robin (WRR) are scheduling algorithms compared by authors. On the other hand, (Mardini et al., 2011) WiMAX technology is based on IEEE 802.16 standard which is a Broadband Wireless Access (BWA) that offers mobile broadband connectivity. But none of them is able to support QoS requirements of the five types of service flow defined by the IEEE 802.16e standard. Some of the past research works uses a history of packet delays to classify packets in four classes and the scheduler gives higher priority to packets destined to users whose instant channel conditions are better. A study on centralized scheduling for Unsolicited Grant Service and Real time Polling Service has been presented by Goyal and Sahoo (2010). The proposed scheduling mechanism meets the quality of service for classes which is discussed by author. Since real time services need extra bandwidth for variable data changing rate, it increases the performance by reducing delay and loss rate. It has been proved that the scheduling algorithm that considered wireless link perform better than the algorithm that does not consider wireless link (Revankar et al., 2010). Chuang et al. (2013) propose a QoS scheme based on Modified Deficit Round Robin (MDRR) of packet scheduling and Call Admission Control (CAC) with the channel condition for nonrealtime service. H.264/AVC is now the standard for video streaming because of its high compression efficiency, robustness against errors and network-friendly features. However, providing the desired quality of service or improving the transmission efficiency for H.264 video transmissions over wireless networks present numbers of challenges. The author (Hsiao et al., 2011) consider those challenges and survey existing mechanisms based on the protocol layers they work on. Finally, they address some open research issues concerning for H.264 video transmission in wireless networks and (Ghazizzadeh et al., 2009) it is estimated according to the instantaneous transmission rate. Fluid Fair Queuing (FFQ) is a well-known algorithm which provides fairness among the packets through the shared link (TCS, 2009). TCS (2009), the author classified the uplink schedulers as Weighted Round Robin (WRR), Earliest Dead line First (EDF) and Weighted Fair Queuing (WFQ). Downlink schedulers are classified into Proportional Fairness (PF), Adaptive Proportional Fairness (APF), Integrated Cross-Layer (ICL) and Round Robin (RR). Revankar et al. (2010), the authors emphasis the MAC scheduling architecture for IEEE 802.16 wireless networks in both uplink and downlink direction to broadcast the frame. Further they used WFQ as uplink as well as downlink scheduling algorithm for improving delay and throughput. There is no separate scheduling policy for Unsolicited Grant Services (UGS). Even though there are vast number of works based on scheduling in single hop
networks, these algorithms cannot be applied for multihop relay scenarios.

**DIFFERENT SCHEDULING Algorithms**

**Round Robin:**
Round Robin as a scheduling algorithm is the most basic and least complex scheduling algorithm. It has a complexity value of $O(1)$ [10]. Basically the algorithm services the backlogged queues in a round robin fashion. Each time the scheduler pointer stops at a particular queue, one packet is dequeued from that queue and then the scheduler pointer goes to the next queue. It distributes channel resources to all the SSs without any priority. The RR scheduler is simple and easy to implement. However, this technique is not suitable for systems with different levels of priority and systems with strongly varying sizes of traffic.

**Weighted Round Robin**
An extension of the RR scheduler, the WRR scheduler, based on static weights. WRR [11] was designed to differentiate flows or queues to enable various service rates. It operates on the same bases of RR scheduling. However, unlike RR, WRR assigns a weight to each queue. The weight of an individual queue is equal to the relative share of the available system bandwidth. This means that, the number of packets dequeued from a queue varies according to the weight assigned to that queue. Consequently, this differentiation enables prioritization among the queues, and thus the SSes. [12]

**Earliest deadline first**
It is a work conserving algorithm originally proposed for real-time applications in wide area networks. The algorithm assigns deadline to each packet and allocates bandwidth to the SS that has the packet with the earliest deadline. Deadlines can be assigned to packets of a SS based on the SS’s maximum delay requirement. The EDF algorithm is suitable for SSs belonging to the UGS and rtVR scheduling services, since SSs in this class have stringent delay requirements. Since SSs belonging to the nrtVR service do not have a delay requirement, the EDF algorithm will schedule packets from these SSs only if there are no packets from SSs of UGS or rtVR class. [13]

**Proportionate Fair Scheduling**
PF was proposed by Qualcomm Company, which was realized in the IS-856 standard for the downlink traffic scheduling (also known as High Data Rate (HDR)) [14]. This algorithm is based on one priority function:

$$Ui(t) = \frac{ri(t)}{Ri(t)}$$

where $ri(t)$ is the current data rate and $Ri(t)$ is an exponentially smoothing average of the service rate received by SS $i$ up to slot $t$. Queue having highest value of $Ui(t)$ is served at time slot $t$. For updating average throughput of the queue following function is used:

$$Ri(t+1) = (1-1/Tc)Ri(t) + (1/Tc)ri(t)$$ if connection $i$ is served at time-slot $t$

$$Ri(t+1) = (1-1/Tc)Ri(t)$$ if connection $i$ is not served at time-slot $t$

Here $Tc$ is time-constant for finding out moving average which is generally taken 1000 slots in CDMA-HDR system [14]. By adjusting this $Tc$ parameter we can make perceived throughput less sensitive to short-term starvation on the queue. So scheduler waits for long time for a particular connection for improvement of its channel quality.

**Cross-Layer Scheduling Algorithm**
To manage resource allocation and grants an appropriate QoS per connection, other scheduling schemes are proposed. These scheduling schemes rely on different algorithms to handle different classes of services for matching their QoS requirements. The algorithm is implemented according to the following formulas. Suppose one frame has $Nd$ time time-slots available. Out of these $Nd$ time-slots, fixed no. of time-slots say Nugs
are allocated for UGS connection. Remaining Nr = Nd–Nugs time slots are allocated to connection having highest priority. Priority function is defined as follows:

\[ Q_i(t) = \beta_{class} \cdot r_i(t) / (R_i(t) \cdot F_i(t)) \text{ if } F_i(t) > 1 \]
\[ Q_i(t) = \beta_{class} \text{ if } F_i(t) \leq 1 \]

\( F_i(t) \) is a satisfaction parameter which is defined as follows:

\[ F_i(t) = T_i(t) - W_i(t) \] for real-time connection, where \( T_i \) is delay requirement specified for connection and \( W_i(t) \) is maximum current delay requirement.

\[ F_i(t) = N_i / n(t) \] for non real-time connection, where \( n(t) \) is data rate specified for connection and \( N_i \) is average data rate which is calculated as follows:

\[ N_i(t+1) = N_i(t)(1-1/T_c) + (1/T_c)r_i(t), \] \( r_i(t) \) is current data rate.

TCP-AWARE UPLINK SCHEDULING ALGORITHM

This algorithm works with only one class of 4 classes defined for QoS. It deals with BE class. As this class has not any specific QoS requirement it is not advantageous to use bandwidth request mechanism for this class and to waste that bandwidth. Also, it is not advisable to equally allocate remaining bandwidth to all remaining BE connections because all connections can’t utilize all bandwidth allocate to them and some may have more requirements than allocated. So, this algorithm works by calculating bandwidth for a particular connection according to sending rate of that connection. Also as sending rate is going to change dynamically, it is not proper to allocate fix amount of bandwidth to a particular connection. So, to properly allocate bandwidth, this algorithm works as follows:

Step 1: Compute the sending rate.
Step 2: If sending rate < allocated bandwidth
Then demand = sending rate
Step 3: If Sending rate = allocated bandwidth
Then demand = increase allocated bandwidth proportionately
Step 4: If sending rate > allocated bandwidth
Then increase bandwidth until sending rate becomes stable. The main strategy of this algorithm is to allocate bandwidth somewhat higher than actual sending rate of connection so that we can safely estimate the sending rate at any given time.

COMPARISON BETWEEN DIFFERENT SCHEDULING ALGORITHMS.
Table 2: Comparison of different scheduling algorithms

<table>
<thead>
<tr>
<th>S.N</th>
<th>Algorithm</th>
<th>Advantage</th>
<th>Disadvantage</th>
</tr>
</thead>
<tbody>
<tr>
<td>1.</td>
<td>Proportional Fair algorithm</td>
<td>Fairness in scheduling</td>
<td>No QoS Guarantee</td>
</tr>
<tr>
<td>2.</td>
<td>Cross-layer Scheduling algorithm</td>
<td>QoS guarantee</td>
<td>Complex implementatio</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Channel quality is considered in</td>
<td>n Slots are allocated to</td>
</tr>
<tr>
<td></td>
<td></td>
<td>scheduling</td>
<td>higher priority connection</td>
</tr>
<tr>
<td>3.</td>
<td>TCP aware uplink scheduling algorithm</td>
<td>Efficient utilization of resources</td>
<td>Complex implementatio</td>
</tr>
<tr>
<td></td>
<td></td>
<td>among BE connection</td>
<td>n handle only one class service</td>
</tr>
<tr>
<td>4.</td>
<td>Cross-layer scheduling for OFDMA networks</td>
<td>Improved packet loss rate, delay</td>
<td>Spectral efficiency of</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>system degrades about 0.3bps/Hz.</td>
</tr>
<tr>
<td>5.</td>
<td>Cross layer downlink scheduling</td>
<td>Scheduling all services flow types</td>
<td>Can be implemented only at the</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Good throughput</td>
<td>base station</td>
</tr>
<tr>
<td></td>
<td></td>
<td>High Frame utilization</td>
<td></td>
</tr>
<tr>
<td>6.</td>
<td>EDF</td>
<td>Focusing on efficiency</td>
<td>Unfit for non real time applications</td>
</tr>
<tr>
<td>7.</td>
<td>WRR</td>
<td>Suitable for non-real time</td>
<td>Does not perform well in variable</td>
</tr>
<tr>
<td></td>
<td></td>
<td>applications</td>
<td>packet size</td>
</tr>
<tr>
<td>8.</td>
<td>Enhanced Cross-layer downlink</td>
<td>Fairness Guarantee to real and</td>
<td>Subscriber mobility is not</td>
</tr>
<tr>
<td></td>
<td>scheduling algorithm</td>
<td>non-real time connection</td>
<td>considered</td>
</tr>
</tbody>
</table>

CONCLUSION

In this paper I have presented various scheduling approaches for satisfying QoS requirements in IEEE 802.16. Algorithms compared are from different approaches so that all available approaches can be covered which can be useful guide for further research in this field. We have tabulated the different parameters on which QoS algorithms can be compared, which will be useful for developing new QoS algorithms. In future I will work on implementing any one algorithm.

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Data Processing
Enhancing Authentication on RFID Tags using CA

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Abstract—RFID is in the limelight for a few years and become pervasive in our daily lives. Security and access control are the most likely applications to use RFID. The biggest challenge for current RFID technology is to provide the necessary benefits while avoiding any threats to the privacy of its users. It is the next generation of bar codes with added functionality and it is rapidly finding more diversified applications in today’s marketplace. This system focuses on the problem of anti-counterfeiting measures that can be provided by RFID tags and investigate which PKC based identification protocols are used for these applications and also to make more secure authentication by enhancing the OKAMOTO protocol through GPS on RFID tags. The Proposed work uses CA (Certification Authority) which avoids DOS attacks. Hence it provides stronger authentication and it uses the CA. Proposed system uses the challenge response and self-certification authority to avoid counterfeiting for each reader and tag.

Keywords—RFID; Hash; Okomoto; GPS; CA; DOS Attacks;

INTRODUCTION

RFID is the acronym for Radio Frequency Identification. RFID technology has been with us for quite some time now. However, rapidly increasing interest in RFID technology is only present in the last few years. The main reasons for this are technical improvements and minimization of cost. For example, RFID technology is now being used for automatic tariff payment in public transport, animal identification and tracking, automated manufacturing. RFID are mainly targeted to provide identification of goods. RFID is an automatic identification technology that utilizes a tag, which may be passive (no internal power) or active (internal battery power) to allow encoded identification, location or other sensory data to be transmitted to a tag reader, which decodes and processes the information.

It also defines non-contact system that can monitor and track items or individuals and it provide unique identification that allows for a wide range of applications. It performs the operation using unobtrusive, low cost components. It uses wireless communication techniques to facilitate the system design. The main goal of an RFID system is to carry data on a transponder (tag) that can be retrieved with a transceiver through a wireless connection. The ability to access information through a non-line-of-sight storage in a tag can be utilized for the identification of goods, locations, animals, and even people. Discerning specific information from these tags will have profound impacts on how individuals in commerce and industry keep track of their goods and each other.

Fig. 1 Block Diagram of RFID

PREVIOUS WORK

Okamoto Protocol,GPS Protocol

This system focuses on the problem of anti-counterfeiting measures that can be provided by RFID tags. It also makes more secure authentication by enhancing the OKAMOTO protocol through GPS on RFID tags.

Okamoto Protocol:
• Online authentication is performed between the tag and the reader and the steps are given below:
• Reader asks the tag for its identification number, ID
• Reader gets from the database a random challenge response pair (ci, xi) for this ID.

- 105 -
- Measures yi.

- Reader verifies whether DL (xi, yi) < §, where § is some predetermined threshold. If this condition satisfied, the reader considers the tag to be authentic, in other case it is decided that this a counterfeit tag.

- The database removes the pair (ci, xi) from the database.

Offline authentication is performed between the tag and the reader and the steps are given below:

Off line authentication is called as PUF-Certificate-Identity based Identification. In this authentication, the readers do not share any secret with the tags. Given a tag with identity I, a PUF, a standard identification scheme SI= (Kg, P, V) where Kg denotes the key generation algorithm, and P, V denote the interactive protocols run by the prover and verifier respectively. Let standard signature scheme SS= (SKg, sign, Vf) with SKg denoting the key generation algorithm, sign denoting the signing algorithm and Vf be the verification algorithm.

For each RFID-tag, having identity I, the issuer then creates a public-secret key pair (pk, sk) using the algorithm Kg. The couple pk, sk is the public secret key pair for the SI scheme. The reader creates the following certificate Cert= (pk, sign (msk, pkII)). The tag sends the certificate Cert to the reader.

Fig. 2 Okamoto protocol

Okamoto’s Identification protocol based on ECDLP

1. Common Input: The set of system parameters in this case consists of (q, FR, a, b, P1, P2, n, h). Here, q specifies the finite field, FR is a field representation, a, b define an elliptic curve, P1 is a point on the curve of order n and h is the cofactor. In the case of tag authentication, these parameters are assumed to be fixed.

2. Prover-Tag input: The prover’s secret (s1, s2) such that Z = -s1P1 - s2P2.

3. Protocol: The protocol involves the exchange of the following messages:

<table>
<thead>
<tr>
<th>Prover P</th>
<th>Verifier V</th>
</tr>
</thead>
<tbody>
<tr>
<td>r1, r2 ∈ Zp</td>
<td></td>
</tr>
<tr>
<td>X ← r1P1 + r2P2</td>
<td>e ∈ Zp</td>
</tr>
<tr>
<td>y1 = r1 + eS1 mod n</td>
<td>y2 = s2</td>
</tr>
<tr>
<td>t = 1, 2</td>
<td></td>
</tr>
<tr>
<td>If y1P1 + y2P2 + eZ = X</td>
<td>Then accept, else reject</td>
</tr>
</tbody>
</table>

Fig. 2 Okamoto protocol

GPS Protocol:

This protocol is public key identification scheme known as GPS due to Girault, Poupard, Stern[3][4], It was standardized within ISO/IEC 9798-5 [5] and it’s also included in the final NESSIE portfolio [6]. GPS protocol is an interactive protocol which contains one or several rounds of three passes between prover and verifier. For this case, the tag plays the role of the prover and the reader is in the role of verifier. It is based on discrete logarithm problem modulo a composite number.

These 3 distinct levels of trust for public key crypto schemes.

Level 1: This level of trust is achieved by schemes were the trusted authority knows or can compute the secret key of any user without being detected. According to Girault [2] identity based public key schemes achieve only this level.

Level 2: This label is achieved by crypto schemes where the trusted authority might not know the user’s private key. Nevertheless the trusted authority can still impersonate a user by providing false guarantees like for example false certificates.

Level 3: Level 3 is achieved by public key Crypto schemes were the trusted authority cannot compute the user’s private key on the one side. And additionally on the other side any attempt of the trusted authority to provide forged guarantees about users is detectable. Certificate based schemes achieve this level of trust.

Fig. 3 GPS Protocol

Problem Definition

Coupons and hash values used in GPS protocol leaks the security of RFID systems.
Even though there are 3 levels of security in the previous work, it gives greater vulnerable to DOS attacks. To overcome these problems, we are going to use CA (called as certification authority) who checks the tag and reader after each and every time the session key generated.

**PROPOSED WORK**

**Security Issue of RFID**
- **Access Violation.**
  The attacker communicates with the tag directly to obtain important information stored in the tag through holding the protocol compatible reader, which result in personal information leaking.

- **Wiretapping:**
  The attacker could detect communication content of the channel between the reader and the tag with RFID device, because the communication between the reader and the tag, the reader and the database is wireless. Thereby, we can capture the content of forward channel (the reader to the tag) and we also could capture the content of backward channel (the tag to the reader) to make doctoring information attacks, replay attacks, counterfeit attacks and so on

- **Doctoring information:**
  The attacker transmits information wiretapped to the receiver after deleting and replacing part or whole of information, which results in the error and invalidation of response message. The purpose of attacks mainly includes malicious destruction of legal tags’ content, prevention of legal tags’ connection establishment and making the receiver believe the message modified is transmitted by a legal user.

- **Counterfeit attack:**
  In the RFID communication network, the connection between the tag and the reader is by means of wireless channel. The tag must transmit its identity information through wireless channel so that the reader could properly identify its identity, but any message could be bugged in wireless channel. After the attacker gained sensitive information of the tag by illegal means, the real tag’s information is copied to the counterfeit tag. When the reader transmits authentication information to the tag, the illegal transmits the tag information copied to the reader so that the tag is in the disguise of legal tag to be passed the reader authentication. Counterfeit attack belonging to active attack is the most popular attack method, which is one of the main hidden troubles faced by system security. The main method to solve the problem about counterfeit attack is the implement of authentication protocol and data encryption.

- **Replay attack:**
  When the reader (the tag) sends authentication information, the attacker captures the response message. When the reader (the tag) sends authorization request next time, the attacker rebroadcasts the previous transmit information of the tag (the reader) to the reader (the tag) so that the tag (the reader) passes the authentication of the reader (the tag) in purpose of acting as the tag (the reader). The attack threatens the RFID system badly, so we prevent the system from being attack with the method of transmission data encryption.

- **Location tracking:**
  The illegal sends fixed information to locate the tag in purpose of tracking and locating. The attacker could send inquiring command anywhere and associate the obtained tag’s certain information with the identity of the tag (the holder) in condition that the tag returns certain information when queried each time. Therefore, the RFID system should satisfy indistinguishability and forward security. Indistinguishability is undistinguishable ability of the information which one tag sends and what others send. Forward security is that the attacker couldn’t verify the tag through obtaining the previous tag sent.

- **Forward security attack:**
  The attacker captures the tag’s output in the communication, and then the previous information sent by the tag could be obtained in relation to current data and history data.
  Because of the limitation of storage capacity and computational capacity of the tag and the reader, the attacker sends a lot of requests to the tag with counterfeit. The tag’s memorizer will halt in that its memorizer stores a lot of random numbers or reads tags to the limit of tag numbers.

**Design of Security Authentication Protocol**

System Initialization process:
Information stored in certificates:
- I. The reader ID \( (R_1,R_2,…, R_n) \)
- II. Certificate value \( (x,y) \)
- III. Session Key \( k \)
- IV. The Tag ID \( (T_1,T_2,…..T_n) \)

Information stored in reader \( (R) \):
- [1] Reader ID\( (R_k) \)
- [2] Certificate value \( (x,y) \)
Authentication process consists of

1. Certification Authority
2. Reader R
3. Tag T

**Certification Authority:**

CA consists of reader information, Tag information, and Session key for reader and tag, Period of validity.

**Steps for Authentication using CA:**

1) **Step 1:** Reader has the initial value of $R_r(\cdot,\cdot)$ and calculates the hash value $\text{Hash}(x \mid R_r(\cdot,\cdot))$.
2) **Step 2:** Tag has the initial value of $T_r(\cdot,\cdot)$ and calculates the $\text{Hash}(x \mid R_r(\cdot,\cdot)) \mid T_r(\cdot,\cdot)$.
3) **Step 3:** Requesting for session key of reader and tag ($S_{kr}, Sk_t$)
4) **Step 4:** Computes $\text{Hash}(x \mid R_r(\cdot,\cdot)) \mid S_{kr}$ for reader
5) **Step 5:** Computes $\text{Hash}(x \mid R_r(\cdot,\cdot)) \mid T_r(\cdot,\cdot) \mid S_{kt}$ for tag
6) **Step 6:** Verify the hashed value of Tag($T_r(\cdot,\cdot)$) and reader ($R_r(\cdot,\cdot)$)

From the comparison, Hash chain protocol, Hash lock protocol and Hash lock protocol couldn’t defend against Dos attacks, replay attacks, counterfeite attacks and internal attacks; they are not secure. ID variation based on Hash still doesn’t solve Dos attack and internal attack; Distributed query-response protocol, David digital library RFID protocol and Tee mutual identify have a good effect on defense eavesdropping, forward security and position trailing, but they couldn’t defend against internal attacks. Nevertheless, the security certificate protocol solves the safety privacy problem of RFID system in the certain degree and it has a good safety.

RFID security protocol not only guarantees privacy and security of information transmission but also synthetically considers the inherent characteristic of the tag and the reader. The characteristic is mainly on the limitation of computational power and memory capacity, which lowers the cost of RFID system. Therefore, we verify the algorithm advantage of the protocol in space complexity and time complexity through the comparison between the protocol and the existing RFID security authentication protocol. The related calculation expressions involved in the protocol are agreed in the following represents Hash calculation; R generates random numbers calculation; XOR represents or calculation; K represents Encryption operation; L represents 128-bit Hash function calculated value.

**LITERATURE SURVEY**

**ANONYMITY (PRIVACY)**

<table>
<thead>
<tr>
<th>Proposal</th>
<th>APPROACH</th>
</tr>
</thead>
<tbody>
<tr>
<td>Inoue &amp; Yasuura</td>
<td>Using two tags – one for unique identification and other for product details, Does not address clandestine</td>
</tr>
</tbody>
</table>
### Table II

<table>
<thead>
<tr>
<th>Proposal</th>
<th>Approach</th>
</tr>
</thead>
<tbody>
<tr>
<td>Juels 2005</td>
<td>PIN: Authenticate the tags to the reader</td>
</tr>
<tr>
<td>Juels 2004</td>
<td>Yoking proofs- Provides cryptographic proofs that two tags were scanned simultaneously and in physical proximity, can be used in a pharmacy to provide to a government agency that the pharmacy scanned a RFID tagged medicine bottle and delivered the exact medicine as prescribes on the RFID tagged prescription.</td>
</tr>
<tr>
<td>Engberg et al. (2004)</td>
<td>Zero-knowledge based protocols for communication between reader tag so they can authenticate each other without revealing any secrets that may allow them to be tracked</td>
</tr>
<tr>
<td>Molnar &amp; Wagner, 2004</td>
<td>Mutual authentication schemes using challenge-response based on the use of pseudo-random function in the computation of response to challenges</td>
</tr>
<tr>
<td>Feldhofer et al., 2004</td>
<td>Proposes the simple authentication and security Layer (SASL) protocol with AES encryption and analyses the hardware requirements.</td>
</tr>
<tr>
<td>Dimitriou, 2005</td>
<td>Provides forward secrecy by using nonces(random number that never reused) by both the reader and tag in their challenges to each other.</td>
</tr>
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</table>

#### CONCLUSION

This project proposes an authentication protocol for security and privacy protection in RFID systems. This protocol protects high-valued goods against attackers. With this authentication, we can provide a proof for each entity of a RFID system.

Our protocol is sufficiently robust to withstand active attacks such as man-in-the-middle attack, the replay attack, the eavesdropping attack and the unwanted tracking of customers. In addition, each tag has its own unique identification data, so user data privacy and location privacy are guaranteed. The Proposed system also focused on the problem of anti-counterfeiting measures that can provided by RFID tags and made more secure authentication by enhancing Okamoto protocol through GPS on RFID tags.
This project also provides reader authentication to a tag, exhibits forgery resistance against a simple copy, and prevents the counterfeiting of RFID tags.

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REFERENCES


Prediction of Next Search Query using Association Rule Mining

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Abstract—The paper presents the method of predicting the next incoming query that the user provides to the search engine. The approach used extracts the information from the log of the previously submitted queries to the search engine, using algorithms for mining association rules. The paper highlights the novel approach of the association rule mining that predicts the next query that the user will provide to the search engine. Using this approach, the search engine keeps the relevant pages in the repository for providing a speedy response to the user and thus increasing the efficiency of the search engine.

Keywords—Search Engine, Query Log, Association rule mining, Apriori.

INTRODUCTION

Web has become an essential source of up-to-date information for the users. Thousands of the users surf the internet to achieve their query results through search engine. Search engine is an information retrieval system [1] designed to minimize the time required to find information over the vast Web consisting of hyperlinked documents. It provides a query interface that enables the users to specify criteria about an item of interest and searches the same from locally maintained databases. The criteria are referred to as a search query. In the case of text search engines, the search query is typically expressed as a set of words that identify the desired concept that one or more documents may contain. The search queries are posed in a different way by every user. There are four types of search queries—Informational Queries[3] that covers a wide topic and gives thousands of relevant answers, Navigational Queries[3] in which these queries are in the form of a single website, Transactional Queries[3] that refer to a particular action, like shopping or downloading a screen saver, Connectivity Queries[3] in which the queries are based on the connectivity of the indexed web graph.

There are various components of Web search engine: crawler module which downloads Web pages, page repository in which the downloaded Web pages are temporarily stored in local storage of search engine, indexing module takes the uncompressed page as input and outputs a compressed version of the page. Another module of the search engine is the query module that converts a user’s natural language query into a language that the search system can understand and consults the various indexes in order to provide a set of relevant pages as answers to the query. The ranking module takes the set of relevant pages and ranks them according to some criteria such as popularity score, content score etc and thus presents sorted results to the user.

The act of the search engine is limited to the problem of “Information Overkill". One of the great challenges faced by the search engines is the difficulty in finding the exact need of the user, as the user enters a short and precise query to the interface. If anyhow the search engine could guess the query that can be posed by the user and what the user wants to know, its performance will definitely improve.

This paper has been organized in following sections: Section 2 describes the current research that has been carried out in this area, Section 3 discusses the proposed work, Section 4 describes the architecture that has been used in the proposed work, Section 5 describes the snapshots of the results of experimental evaluation, Section 6 concludes the discussion.

RELATED WORK

Fonseca, Golgher, De Moura, and Ziviani in [2] segmented query sessions in search engine query logs into subsessions and then used association rules to extract related queries from those subsessions. Association rules are widely used to develop high-quality recommendation systems in e-commerce applications available in the Web (Agrawal, Imielsinski, & Swami, 1993; Agrawal & Srikan, 1994). These applications take user sessions stored in system logs to obtain information about the user behavior to recommend services and products.
In [3], a novel approach to predict the oncoming query for the search engine has been discussed. This approach uses neural networks for the prediction.

In [4], the authors predict users' future queries and URL clicks based on their current access behaviors and global users' query logs. They also explored various features from queries and clicked URLs in the users' current search sessions, select similar intents from query logs, and use them for prediction. Query Log excerpt (RFP 2006 dataset) has been taken as an experimental corpus. Three methods and the back-off models have been presented.

In [5], a method to help a user redefine a query based on past users experience, namely the click-through data as recorded by a search engine has been presented. The method proposed attempts to recommend better queries rather than related queries. It is effective at identifying query specialization or sub-topics because it takes into account the co-occurrence of documents in individual query sessions.

**PROPOSED WORK**

The paper presents a novel approach to predict the next query that the user may provide to the search engine. The approach discussed uses association rule mining. Given an initial input query, the user can see the list of the queries which he may be interested in posing next to the search engine. For the purpose, the system for query prediction has been proposed.

The proposed system of query prediction uses the following steps:

**Step-1 Maintaining the Query Log**

First, the proposed system maintains a query log for the user. Query log consists of transactionId, date, time, queryId and name of the queries entered by the user. The transactions in the Query log are maintained on the day to day basis.

A *query log* is basically a file containing the interactions that occurred between the users and the system. The content of the search engine’s web server log depends on the server and its settings. The log’s entries can be simple or complex and presented in different forms. But in general, the log contains the following information:

- Time and date of transaction.
- Name of the query.
- IP address to which the query was sent.
- How the query was sent?

Possible link that led to that page.
Information about the browser.
Transmission results.
Rank of the clicked result.

The following is an example of an entry in the Query log.

1042078585.991                      3713
200.226.211.142
TCP MISS/200        25368      GET
http://cluster.igbusca-cluster/query.cgi?
query=+origem+da+familia+marques-
DIRECT/192.168..12 text/html

**Step-2 Applying the Query Mining System on the Query Log**

Now, the query logs of different users, their descriptions, interests and the queries they usually pose to the system at a particular time has been maintained. Section IV describes the proposed “Query Mining System” that can be applied on the query log to predict next the incoming queries to the system.

**Step-3 Predicting the queries**

Since, each user will have different query log, the user login will be required to identify which user has logged in. The system maintains search interface in which the user will provide the initial query to the. And there is another search box; in which the user will provide the minimum support (frequency) count. Whenever user clicks the button of the related queries, the user will get a list of the next predicted queries. In this way the system, helps the particular user logged in, to get the list of the next queries to help to tune the search process. It not only fastens the search but also increases the efficiency of the system, as it will store the next predicted queries in its cache to provide the desired result page to the user in a very less time.

Thus, Query Prediction is one of the major functionality required for the search engine. Thus, the Query prediction system can be used in the search engine’s architecture to predict the next query provided to the search engine by the user on the basis of past queries entered by him.

There are two key performance indicators for the web search engines:

1. Quality of the returned results
2. Speed with which the results are returned.
Query prediction aims to increase the speed with which the search engines respond to the users with the returned results i.e. the response time of the search engine will reduce and thus its efficiency will improve.

**QUERY MINING SYSTEM**

![Figure1: Process employed in Query Mining System](image)

The process employed in Query mining system consists of two main functionalities –one for extracting user sessions and other for mining association rules.

4. Extracting user sessions

User session is basically for how much time the user is active on the server. In our proposed system, the time field is divided into time intervals of say t minutes each and the user session is defined by the set of queries each user (identified by the IP address) submits in the time interval. If many queries are submitted within the time interval, then the system restricts the number to a minimum value.

5. Mining Association rules

Association rule mining is one of the data mining techniques which aim to discover the interesting patterns from the large transactional databases. For the system, the technique is applied on the query log that has been maintained. An association rule is identified as the rule X=>Y where X and Y are set of the items. In our system, X and Y are the queries which are divided into the set of transactions. In the association rule mining, we first do the frequent pattern analysis in which we find the patterns (set of items, subsequences, substructures etc.) that occur frequently in the set.

This can be defined mathematically way as let \( 1=\{I_1, I_2, \ldots, I_n\} \) be a set of literals called items. Let \( T \) be the database of transactions. Each transaction \( t \) can be represented by a binary vector, with \( t[k]=1 \) if \( t \) bought the item \( I_k \), and \( t[k]=0 \) otherwise. Let \( X \) be a subset of \( T \). A transaction \( t \) satisfies \( X \) if for all items \( I_k \) in \( X \), \( t[k]=1 \).

Similarly, in this way, we can define our problem of finding next predicted queries. Here, the set \( Q=\{Q_1, Q_2,\ldots, Q_n\} \) be a set of queries from the query log database. Let \( T \) be the database of set of user sessions. Each session \( t \) can be represented by a binary vector, with \( t[k]=1 \) if query \( Q_k \) fired in session \( t \), and \( t[k]=0 \) otherwise. Let \( X \) be a subset of \( T \). A transaction \( t \) satisfies \( X \) if for all queries \( Q_k \) in \( X \), \( t[k]=1 \).

There are two important parameters that we need to consider.

V. **Support**

The rule \( X=>Y \) has a support factor of \( s \) if \( s\% \) of the transactions in \( T \) contains \( X \cup Y \). The problem of mining association rules is to generate all the association rules that have a support greater than a minimum support threshold (\( \text{minsup} \)) set up during the experimental analysis. Thus it shows only those rules in which all queries together appear more number of times than the number set up that is \( \text{minsup} \).

[7] **Confidence**

The rule \( X=>Y \) has a confidence factor of \( c \) if \( c\% \) of the transactions in \( T \) that contains \( X \) also contains \( Y \). This means the rule like \( Q_1=>Q_2 \) has a confidence factor \( c \) if in \( c\% \) of the total transactions in database, if \( Q_1 \) occurs then \( Q_2 \) also should also occur.

The frequent patterns or rules will be output if they are having support and confidence greater than or equal to the minimum support and confidence set up during the experimental analysis.

There are many methods to do association rule mining like apriori algorithm, partitioning method and FP growth method.

We have used the apriori algorithm in this paper. In apriori algorithm, we find the large frequency itemsets. In the first pass, we count the item occurrences to determine the large 1-itemsets. A subsequent pass, say pass \( k \), consists of two phases. First, the large itemsets \( L_{k-1} \) found in the \((k-1)\)th pass are used to generate the candidate itemsets \( C_k \) using the apriori-gen function. Next, the database is scanned and support of the candidates in \( C_k \) is counted. The apriori-gen function takes an argument \( L_{k-1} \), the set of all large \((k-1)\) itemsets. It returns a...
superset of the set of all large k-itemsets. It performs two functions. First in the join step, we join $L_{k-1}$ with $L_{k-1}$. Next, in the prune step, we delete all itemsets $c \in C_k$ such that some $(k-1)$ subset of $c$ is not in $L_{k-1}$.

6. Predicted Queries

After mining the association rules, the queries that occur with the current query in the rules will be the output. This is because as per the rules the query in the output will have maximum probability to be fired by the user next to the current query. Also, the association rules go on updating periodically and the relation between the queries can be found to predict the next query.

EXPERIMENTAL EVALUATION

The proposed system has been implemented in Java. The following figures (Figure 3 to Figure 7) show the snapshots of the implementation.

3. Query Log is maintained for each user as shown in the Figure 3.

4. Association rule Mining is applied on the query log maintained as shown and frequent querysets are found.

5. After finding the frequent querysets, now the system finds the queries having maximum probability and which the user can fire next to his current query on the interface.

Figure 3: Snapshot showing how Association rules are mined and frequent query sets is found

Figure 4: Snapshot showing how the next query is predicted
While implementation and experimental evaluation, a minimum threshold has been set for the support and all the queries that has the support value greater than or equal to the decided threshold value will be considered.

4. CONCLUSION

Association rule discovery is one of the most important techniques in the field of data mining. It aims at finding patterns that exists in the databases. Using the approach of Association rule mining, the proposed system works for the prediction of the query that the user is more likely to fire next to his current query. The proposed system worked well for some sample queries entered by the user and has shown promising results. The same approach can be used in search engine to improve its efficiency and thus providing better results to the user.

5. REFERENCES

Summarization of Search Results Based On Concept Segmentation

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Abstract—The main objective of Summarization of Search Results based on concept segmentation (SSRCS) is to provide the comparative summary of the multiple URLs selected by the user based on the search query, uses an algorithm known as Concept Based Segmentation [1]. SSRCS uses HTML DOM tree structure of the web pages. The HTML DOM tree of each web page of the URL is formed and is segmented to form the concept blocks. The sentence score of each of the concept block is evaluated based on the search query and useful information is extracted to generate a comparative summary. This method do not uses any preprocessed data and all the actions are performed dynamically to support a wide range of documents. This system will help the user in terms of saving the time and effort required. Moreover, the user will be able to take a decision in a very less time.

Index Terms—Concept Based Segmentation, HTML DOM, Comparative Summary.

1. INTRODUCTION

A The huge growth of World Wide Web (WWW) with technological growth is serving million of users. The tool called search engine is used to search the information from world wide web (WWW) and generates millions of results based on a query. But the users still have to go through the various results produced to find the required information. The current search engines produces a small summary with each result which usually contains the first few lines of the web page or the lines containing the keywords based on the query.

This paper is an extension of the work mentioned in [1]. This paper proposes a dynamic comparative summarizer which extracts the important sentences from the user selected URL’s. This extraction of important sentences depends upon the user fired query. This will help the end user in saving the time and effort required for quick decision making. The rest of the paper is organized as :- Section 2 describes the Related Work and Section 3 explains Problem Formulation for this paper. Section 4 explains the proposed work. Section 5 describes the experimental setup and their results are described in Section 6. Section 7 concludes the paper along with the future work mentioned in section 8.

2. RELATED WORKS

Summarization of web documents is an extensive area for research and various related works has been done in the past.

An Automatic Text Summarization Approach using Content Based and Graph based characteristics was proposed in [2]. In the first stage, the content-based vectors are used to represent the segments followed by the compression of these vectors into lower dimensional matrix in order to find out the hidden patterns. In the second stage, the nodes were used to represent the segments and the edges were used to represent the relationship between two segments having the similarity score above the threshold. Then a graph search technique was used to determine the segments to be included in summary.

Ahmed A. Mohamed et. al. proposed a technique in [3] for improving Query Based Summarization using Document Graph. This technique focused on generating a user-focused summary based on a query. The document graph for all the sentences in the input documents was generated. Then the document graph for the query was generated followed by measuring the similarity between each sentence and the query. The best sentence was searched and was added to the summary until the limit of the summary length is reached.

The study of [4] proposed a technique for Organization of Documents for Multiple Document Summarization which can be used in fast growing e-learning. The hierarchical summarization models based on the hierarchical structures of a document provide an effective summarization technique. Three hierarchical structures are proposed to organize a collection of news stories. In the first structure, the news stories are stored into a hierarchical tree in chronological order and the organized as balanced hierarchical tree. Hierarchical summarization extracts the information distributively. In the second structure, the news stories were organized into hierarchical structures by time interval such that each child node represents equal and non overlapping

- 116 -
interval. It was an unbalanced tree structure. Therefore information was non evenly distributed. The third structure, organize the news stories into hierarchical structures by event topics because the accuracy of event topic detection affects the performance of summarization directly. Md.Mohsin Ali et. al. proposed two techniques for both single and multi document text summarization i.e the CPSL technique and the LESM technique [5]. CPSL technique was the combination of SimWithFirst and MEAD. MEAD was a centroid-based extractive summarizer. The sentences were scored based on some sentence features such as centroid, position and length. The high scored sentences were included in the summary. In SimWithFirst, the similarity score between current and first sentence was evaluated and the highest scored sentence was taken as the most similar sentence. Then the cosine similarity between a sentence and the first sentence was evaluated. This was further combined with MEAD determine which sentence to include in the summary on the basis of sentence score. First, the summary according to LEAD and CPSL is determined. Then, the common sentences from the two summaries are extracted. Then, on the unmatched sentences the LEAD based technique was applied for obtaining the desired amount of summary.

Chen and Jie proposed a technique in [6] for query based automatic summarization of web page. The segmentation of HTML documents into topic blocks was carried out. Then, for each sentence in the topic block, the weight is evaluated based on query keyword. Then, the extraction of important sentences was performed dynamically to obtain the summary of expected compression ratio with the maximal marginal relevance method which removes the redundant information in the resultant summary.

The idea of using Semantic Role Labeling (SRL) was proposed in [7] which work on multi document summarization (MDS) [7]. It proposed a method SRLSUM. According to this method, the documents were first given as input to the SRL parser. Then, the semantic arguments were extracted from all the parsed sentences. The stemming of the argumented and the original documents was performed to remove the stop words. The frequencies of every semantic argument were evaluated for a particular topic. Then, these arguments were used to score each sentence. The top scoring sentences were included in the summary and the redundancy is also eliminated.

This research work focuses on the concept of generating the comparative summary of the documents by aggregating the document summaries. This makes the use of concept based segmentation of HTML DOM tree structures of the web pages. The comparative summary is composed of the important sentences related to the query which are extracted from the concept blocks of different web pages. This system focuses on the dynamic creation of HTML DOM tree and the concept blocks which are utilized for summary generation.

3. PROBLEM FORMULATION

The earlier work done in this field mentioned in [1] has some drawbacks. The very first problem is the static database. It uses preprocessed data. The creation of DOM tree and the concept blocks are all done prior to run time for a specific set of data only and generate a comparative summary for the same which diminishes the importance of the method. The next drawback is, a large database requirement for storing the preprocessed data and hence a large amount of memory space will use which will decrease the efficiency of the system and increases the cost of the system. Moreover, the current system also suffers from network congestion and bandwidth wastage which increases the burden on the system. Bandwidth wastage is due to unnecessary data download which is not used further.

4. PROPOSED WORK

This paper proposes the use of dynamic database for the generation of comparative summary. All the work from the generation of HTML DOM tree to the formation of concept block which has been done earlier using a static database is now implemented dynamically. All the preprocessing is now implemented at run time only. This removes the need of a static database. Also, there is no need to have a large amount of database as all the records from the database are removed after generating the comparative summary. Moreover, this proposed method will work for all sets of data and not for specific set of data.

4.1 Proposed and Implemented Architecture

This system uses an algorithm mentioned in [1] known as Concept Based Segmentation. Concept based segmentation is the technique in which the HTML documents are segmented into concept blocks. The concept blocks contain the actual content of the document. The important sentences from these
concept blocks are extracted based on the search query to generate a comparative summary. Architecture and working of proposed system is shown in figure 1.

One need to do various things prior to perform concept based segmentation. The following things have been performed to perform Concept Based Segmentation:

4.2 Search Engine

The very first thing needed is a search engine. Since this paper focuses on comparative summary generation and not on the creation of search engine therefore we do not specify the method for the same. However, this system uses a Bing search engine API. Bing search engine API enables us to embed a
flexible and powerful search engine as a custom search component in websites and applications. In order to use the API first we need to get an AppID which can be easily obtained from Bing developer center website URL.

The search engine apart from the textbox for writing the query and a search button must contain checkboxes on the left hand side of each URL and a compare button. For the inclusion of checkbox and compare button we need to modify the API. The checkbox facilitates the user to select URLs for which user wants to generate the comparative summary. Compare button is used to generate comparative summary of the selected URLs.

4.3 HTML DOM Tree and Internal Links
The next step is to extract the source code of the web document and to create the HTML DOM tree for the same. Moreover, we also need to find out all the internal links present on the web page because the information related to a query is not necessarily available on the single web page and may be widespread throughout the different internal links on different web pages. Therefore, the web document for all the internal links must be extracted and the HTML DOM tree for all the internal links must also be created.

4.4 Topic Block
The Document Object Model (DOM) is used to analyze the content of the web page. The leaf nodes of DOM tree also known as micro blocks contain the actual content of the web page whereas the parent nodes generally contain the higher level topics. Adjacent micro blocks of the same parent tags are merged to form topic blocks.

4.5 Concept Block
Topic blocks containing the information about the similar concept word are merged to form the concept block. Concept based similarity between the topic blocks are measured using a semantic role labeler (SRL) to identify the similar topic blocks. WordNet is used as SRL to identify the similar topic blocks. It compares each topic block with other topic blocks and based on the comparison it generates a similarity score. The topic blocks having the similarity score above the threshold value $\alpha$ (0.9) are merged to form concept block. Moreover, the web documents extracted are filtered by removing the unwanted HTML tags (like META tag, ALIGN tag, CSS style tags etc.). Moreover, the tags like &nbsp has been replaced by space character as these tags will not contribute for summary generation. After the formation of concept blocks, the important sentences are extracted from the concept blocks. Further based on the search query for multiple URLs selected by the user is used to generate comparative summary.

5. EXPERIMENTAL SET UP
For carrying out the experiment we have used various tools and technologies. We have used Microsoft Visual Studio as the IDE for the development of the framework in C# language. We have used ASP.NET technology for the implementation of the framework. Apart from this we have used HTML Agility pack for the creation of HTML DOM tree. For the installation of HTML Agility pack we require NUGET software. It is a collection of tools to automate the process of downloading, installing, upgrading, configuring, and removing packages from a VS Project. Moreover, we have used WordNet software which is a large lexical database of English. Nouns, verbs, adjectives and adverbs are grouped into sets of cognitive synonyms (synsets), each expressing a distinct concept. It compares each topic block with other topic blocks and based on the comparison it generates a similarity score. The topic blocks having the similarity score above the threshold value are merged to form concept block.

6. EXPERIMENTAL RESULTS
The experiment is carried out on real time data set from the internet. It works fine for very small website which has very less number of internal links. However, for large websites it takes some time to generate the comparative summary because a large website contains a large number of internal links. When a user clicks on a URL of such large website, all the internal links of the URL are also being hit simultaneously in background for the extraction of web document and creation of HTML DOM tree which increases the hit ratio and hence slow down the process of summary generation. Figure 2 and figure 3 shows the results of the proposed system.
CONCLUSION

We have implemented the framework called SSRCS in C# language. The results achieved from
the proposed framework is better than the existing system i.e. the drawbacks like static database, Bandwidth wastage, memory wastage, network congestion has been removed. This system focuses on the generation of comparative summary dynamically. The generation of DOM tree and the creation of concept block are done at run time only which removes the need of a static database and saves a lot of memory space needed for storing the contents. Moreover, it works for all sets of data. The important sentences based on query are extracted from concept blocks and are merged to form comparative summary. The final results of the project also proved that the searching and analyzing time of the user is reduced significantly. The comparison of different text summarizers are provided in table1.

### 7. FUTURE WORK

This system utilizes the content present within the `<p>` tag only for summary generation. The inclusion of contents from other tags in summary generation can be done as future work. Apart from text different types of documents like images, audio etc. can also be considered to generate their comparative summary. Moreover, the presentation of summary can be improved in order to determine the comparisons more effectively.

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Search Engine Snippets</th>
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<th>Comparative Summarizer</th>
<th>SSRCS</th>
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</thead>
<tbody>
<tr>
<td>Method of summary Generation</td>
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<td>Extraction</td>
<td>Extraction</td>
<td>Extraction</td>
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<td>Techniques used</td>
<td>Occurrence of query string</td>
<td>Lex rank, Centroid Position</td>
<td>DOM tree, Concept based segmentation</td>
<td>DOM tree, Concept based segmentation</td>
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<td>More Control</td>
<td>More Control</td>
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<tr>
<td>Satisfaction Index</td>
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<td>Medium</td>
<td>Medium to high</td>
<td>Medium</td>
</tr>
<tr>
<td>Run time over head</td>
<td>Less</td>
<td>Comparatively High</td>
<td>Comparatively high</td>
<td>Comparatively High</td>
</tr>
<tr>
<td>Usage</td>
<td>Gives a clue about the relevance of the document</td>
<td>Short summary is generated</td>
<td>Comparative Summary is generated which is useful for decision making</td>
<td>Comparative Summary is generated which is useful for decision making</td>
</tr>
<tr>
<td>Retrieval Efficiency</td>
<td>Need further browsing and Scanning</td>
<td>Reduces time taken for scanning entire set of documents to understand the core concept</td>
<td>Reduces time and effort taken for browsing and scanning various web pages to extract the gist of it</td>
<td>Reduces time and effort taken for browsing and scanning various web pages to extract the gist of it</td>
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<td>Database</td>
<td>Static Database</td>
<td>Static Database</td>
<td>Static Database</td>
<td>Dynamic Database</td>
</tr>
</tbody>
</table>
REFERENCES


Sentiment Analysis based Approach for Interlinking Events to Analyze the Performance of Government during Disaster

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Abstract- This paper proposes a method for linking various disaster related events such as damage/loss effects, recovery effort and the people feedback. The main idea is to generate a linking algorithm using sentiment analysis based approach through which we can analyze the performance of the government recovery effort during the disaster. The linking algorithm takes the input in the form of different data sets describing news regarding damage/loss effects, recovery type events and the people feedback for each disaster. SentiWordNet and manually created vocabulary related to disaster is used for linking various disaster related events. We used threshold based risk table for evaluating people feedback score and to determine the performance of government during the disaster to check whether the recovery effort is appropriate during the time of disaster or not so that it will helpful to analyze the government effort and to have better resource planning for future.

Keywords- Sentiment analysis; Event linking; K-nearest neighbour Text mining; Performance Analysis.

INTRODUCTION

A disaster can be defined as any occurrence that cause damage, ecological disruption, loss of human life, deterioration of health and health services on a scale, sufficient to warrant an extraordinary response from outside the affected community or area. Disasters are mainly of two types: i) Natural Disaster example - earthquake, floods, landslides etc. ii) Man-made Disaster example - war, bomb blast etc.

Requirement for developing an automatic system using sentiment analysis based approach for linking disaster related events: Human evaluation for measuring the performance of disaster related recovery efforts is very lengthy and time consuming so to minimize the effort and time for measuring the performance of recovery effort we focus our research work towards the development of an automatic system through which we can analyze the performance of government recovery effort with respect to damage/loss occurred during the disaster. For this we have used sentiment analysis technique.

Sentiment analysis is the process of classifying the polarity of a given text in documents, sentence, or feature/aspect level whether the expressed opinion in a document, sentence or an entity feature/aspect is positive, negative, or neutral.

Flow of our approach in developing an automatic system: We collected news regarding damage/loss effects, government recovery effort, people feedback for each disaster from various online news channels and formed different datasets related to each disaster events and then using sentiword net and manually created library we calculate the interlinking among various disaster related events and then based upon the total scores that we have calculated for each disaster related event we can analyze the performance of government recovery effort during the disaster.

A. Our Contribution in developing an automatic system

We have developed a new approach for interlinking various disaster related events and our automatic system will analyze the
performance recovery effort applied by the government or related organizations with respect to damage/loss and people feedback. Our contribution in developing an entire system can be listed below:

- We formed different datasets related to disaster describing the news regarding damage/loss, government recovery effort, and people feedback for each disaster.
- We have developed a new approach for linking various disaster related events for various datasets.
- Finally, we have develop a complete system, which ranks the recovery effort and divide it into fixed set of remarkable categories, like: good, average, bad, very bad etc. to determine the performance of government recovery effort for the disaster.

**B. Paper Organization**

In Section 2, we discuss some Related Works nearest to our research area. In Algorithm using sentiment analysis based approach for linking disaster related events, we generate an event linking algorithm for solving the problem (Section 3). In Pseudo Code, we define step by step working of linking algorithm (Section 4). In Evaluation, we define the details of the data set that we have used, experimental settings required for the system, we have used an evaluation metrics to measure the performance of the retrieved system and experimental results show the result of the system that we have developed (Section 5). Finally, we concluded our research work in section 6.

**I. RELATED WORK**

Ahmed Nagy and Jeannie Stamberger [1] introduced a new approach for scavenging critical information through micro blogs, which can be used to detect the sentiments of the crowd towards crises or disaster which uses two approaches annotated word lists and classifiers.

Benjamin, Aron, John , Danielle, Bonnie, Jeremy [2] performed a demographic analysis of online sentiment during Hurricane Irene which reveals the differences in the sentiment depending on the person’s gender or location and concluded that social media analysis provides a viable, real-time complement to traditional survey methods for understanding public perception towards an impending disaster.

Two state-of-the-art SVM classifiers[3] was created in which one of them is used to detect the sentiment of messages such as tweets and SMS (message-level task) and other to detect the sentiment of a term within a message (term-level task).

Survey research by West and Orr(2007) concluded that women may feel more vulnerable during hurricanes because they are more likely to have children and belong to a lower socio-economic class. Richer people tend to have an easier time dealing with natural disasters like hurricanes. West and Orr also finds the differences in regional perceptions of vulnerability between coastal areas and non-coastal areas. Tan Lee, Tang, Jiang, Zhou, Li (2011) presented an approach where they took into consideration the sentiment of a user’s social network to calculate the overall sentiment. Nielsen (2011) presented a list-based approach to calculate the Tweet sentiment.

Barbosa and Feng (2010) introduced a classification method and concluded that the top five features in terms of information gain are: negative polarity, positive polarity, verbs, emoticons and upper case and applied a summation and normalization approach to calculate the sentiment of Tweets gathered from twendz, twitter sentiment and found that the positive words awesome, rock, love and beat express 95% of the positive sentiment while 96% of the negative sentiment is expressed by the words hate, suck, piss, stupid and fail.

**II. ALGORITHM USING SENTIMENT ANALYSIS BASED APPROACH FOR LINKING DISASTER RELATED EVENTS**

In this section we discuss a linking algorithm, which is used to link different disaster related threads such as damage and loss caused by disaster, recovery effort applied by government /related organization and people’s feedback, spread in different online news sources available. Finally, we calculate the impact of recovery efforts based on some evaluation category. The steps are given below:
A. Collecting disaster related news threads and forming the datasets related to each disaster event

We collected the documents that we have retrieved through online news articles for each disaster and forming datasets describing the news regarding damage/loss, news regarding government and private effort, and the news describing the people feedback respectively for each disaster event.

B. Linking of various disaster related events

In this step we perform linking between various disasters related events i.e. the datasets describing the damage/loss, recovery type events and the people’s feedback which includes further processing steps.

- Preprocessing: We apply a simple preprocessing step to clean the dataset. The preprocessing steps include input cleaning such as stop words removal and tokenization of documents. For input cleaning, we remove the stop words, regular expressions, semicolon, commas, and inverted commas from the corpus. Finally, we get the dataset that consists of only dictionary meaning words separated by blank space. In this step we match each and every word present in the dataset with the list of stopword if there is a matching then it will remove/replace that matched word from the dataset.

Note: A stop list is a set of words that are deemed “irrelevant”. For example a, an, the, of, for, etc. are stop words.

- Event Linking : To link the common events in three different fields, i.e. (1) disaster related story lines, (2) recovery efforts for corresponding disaster and (3) user’s/ people’s feedback on recovery effort, we use the density of common keywords. For this we manually create the vocabulary for corresponding disaster, recovery efforts and the user’s feedback.

Now the main task is to identify the impact of various disasters related terms (obtained through our vocabulary). As, sometime in a particular disaster, some terms may have more impact than others. For example, in the case of an earthquake, the destruction of houses, road, etc. will have more impact than diseases, etc. So, to calculate the impact of such disaster related terms, we use the importance score of disaster related words calculated by using “tf-idf” score and K-nearest neighbour weighted words.

Identifying important words which represent particular disaster, recovery effort and user’s feedback: For this we use our vocabulary and identify all words related to disaster, recovery, and user’s feedback present in each dataset. Next, we apply the following ranking approach to identify the matching words related to all the above discussed events:

\[ Score_{imp}(W_i) = tfidf(W_i) \times Weight_{Knn}(W_i) \]  (1)

Where, \( Score_{imp}(W_i) \) = Importance score of word \( W_i \) in document. \( tfidf(W_i) \) = tfidf score of word(\( W_i \)). \( Weight_{Knn}(W_i) \) = Score of highest weight K-nearest neighbour word present near to given word \( W_i \). The tfidf score of any word can be calculated by using:

\[ tfidf(W_i) = tf(W_i) \times IDF(W_i) \]  (2)

Where, \( tf(W_i) \) = Term frequency of word \( W_i \) in given document. which is given by the formula:

\[ tf(W_i) = 1 + \log(1 + \log(freq(W_i))) \]  (3)

where \( freq(W_i) \) is the number of occurrences of the word in the document. If “D” is the total number of documents in corpus and "D, " is the number of documents containing the word \( W_i \).

Then \( IDF(W_i) \) can be calculated as:

\[ IDF(W_i) = \log \frac{1 + D}{D_i} \]  (4)

Event Linking: To link the events, i.e., (1) disaster, (2) recovery effort and (3) feedback, we identify the presence of density of top ranked words related to each event. For example, to identify the disaster related news story, we select the paragraph having the highest density of top ranked disaster related words. Next, to identify the recovery efforts for the same disaster, we identify the paragraph having “top ranked recover effort related words” and “top ranked corresponding disaster related words”. Finally, to identify the feedback related news segments, we identify the paragraphs having a high density of (1) top scored feedback related words and (2) high density of the corresponding top ranked disaster related words and (3) high density of
corresponding recovery efforts related top ranked words.

Note: The final outcome of this approach is separate news segments, related to the disaster, recovery for that disaster and feedback. Here we consider news paragraphs as story segments.

C. Calculating the impact of recovery effort based on disaster, recovery and feedback

For this we use automatically, extracted news segments related to disaster, recovery efforts and feedback. Next, we use the sentiment score of words and their density in calculation of level of recovery efforts.

To calculate the effective sentiment of disaster related news threads, we take all the words which show negative sentiments. Actually, we want to capture the notion of loss/damage in this case and this is the main aim behind the selection of negative sentiment KNN weighted words. The Score for disaster related news $T_{\text{disaster}}$ can be given as:

$$T_{\text{disaster}} = \sum_{i=1}^{n} \text{Score}_{\text{imp}}(W_i)$$

(5)

Where, ‘n’ total number of damage/loss related words that matches with the vocabulary in the disaster related news threads.

Next to calculate the effective sentiment of recovery related effort applied by recovery agencies, we select all positive sentiment KNN weighted words from news thread related to efforts applied by disaster recovery agencies. The utilization of positive cases in the calculation, is the main aim behind this selection. Now the score related to recovery effort $T_{\text{recovery}}$ can be given as: (here, ‘r’ is the total number of recovery related words that matches with the vocabulary in news threads related to recovery efforts).

$$T_{\text{recovery}} = \sum_{i=1}^{r} \text{Score}_{\text{imp}}(W_i)$$

(6)

Finally, to calculate the effective sentiment score of user’s feedback related events, we take all sentiment KNN weighted words from news threads related to feedback. The main aim is to identify the lack in applied resources and efforts.

$$T_{\text{feedback}} = \sum_{i=1}^{f} \text{Score}_{\text{imp}}(W_i)$$

(7)

Where, $T_{\text{feedback}}$ represents the effective sentiment of feedback related events. ‘f’ is the total number of feedback related words that matches with the vocabulary present in the recovery related news threads. Finally to rank the recovery efforts applied by recovery agencies, we take the difference between (1) $T_{\text{disaster}}$ and $T_{\text{feedback}}$ and (2) $T_{\text{disaster}}$ and $T_{\text{recovery}}$. Let $A = T_{\text{disaster}} - T_{\text{feedback}}$ and $B = T_{\text{disaster}} - T_{\text{recovery}}$.

Now, to get the level of satisfaction by recovery efforts, we calculate the percentage score of ‘B’ w.r.t., ‘A’. Let us represent it by ‘P’.

**Ranking the disaster related recovery efforts:**

We use the value of ‘P’ to rank the efforts. If P is less than 20%, then it indicates worst performance (also represented as -2), if ‘P’ lies in the range (20%,40%) then it indicates bad performance (also represented as -1). Next if ‘P’ lies in the range (40%, 60%) then it indicates average effort (also represented as 0),If ‘P’ lies in the range (60%, 80%) then it indicates good effort (also represented as +1),and finally if ‘P’ shows value greater than 80% then it indicates best effort towards disaster recovery (also represented as +2). All the percentage values are converted into the range of [0,1] in all experimental results.

### III. PSEUDO CODE

**Input:** The input will be the collection of online news articles obtained by crawling the web with some restricted keywords.

**Output:** The output will be the set of interlinked events among disaster related news describing the level of recovery efforts applied by the recovery agencies.

**Algorithm using sentiment analysis based approach for linking disaster related events :**

Step I Collection of disaster related news threads through online news articles by forming different datasets related to each disaster.

Step II Apply simple preprocessing and input cleaning(See Sub-section ‘A’ of section III).

Step III Use the collected news stories and identify the links between various disasters related news event threads.

Here, interlinked news events include:
(1) Effect (loss/damage) due to disaster, (2) recovery effort applied by recovery agencies and (3) people’s feedback related to the recovery effort. (See subsec- tion 'B' of section III).

StepIV Use interlinked events to calculate the impact of recovery effort based on the disaster, recovery and feedback (See Sub-section 'C' of Section III).

StepV Apply Step1 to Step 3 for all different disasters related news story collections.

IV. EVALUATION

Entire System is developed in C#.Net (Microsoft Visual Studio 2005 on Windows XP. To find the relationship between various disasters related events we used linking algorithm for linking disaster related events.

A. Aim and Goal Of Our System

On the basis of the datasets which we have collected from the web and using linking algorithm our system has generated an automatic level of feedback which evaluates the performance of the government effort through people’s feedback with respect to the damage/loss occurred during the disaster to check whether the recovery system of the government/recovery agencies is appropriate or not.

B. Details of Dataset used:

- Web based datasets: We have collected the details of five days of data from when a disaster has occurred from the web. Here we analyze the interlinking among various disasters related events for various disasters such as Earthquake that occurred in Sikkim on 18th September, 2011, Flood that occurred in Alberta on 20th June, 2013, and the Cyclone that occurred in Phailin from various sources such as online news articles, CNN and BBC News etc.
- Our Own Data set constructed by using publically available Text data sources: We collected 150 documents related to the disaster earthquake occurred in Sikkim out of which 140 are relevant, 170 documents for the disaster Flood occurred in Alberta out of which 160 are relevant, 126 documents for Phailin Cyclone out of which 126 are relevant.

C. Experimental Settings

We use an Event linking algorithm that generates the score for each datasets related to disaster and based upon the scores we interlink various disasters related events. A threshold based risk table is used in which we divide the threshold into different ranges of risk level as well as the type of risk associated with it which is used for evaluation. We use an evaluation metric such as precision, recall and f-measure to evaluate the performance of the information retrieved from the web.

D. Evaluation metrics

We use precision, recall and f-measure to evaluate the performance of documents that we have retrieved from the web.

If $R_{Doc}$ is the total number of retrieved documents from the web, $REL_{Doc}$ is the total number of relevant documents, $P$ represents the precision and $R$ represents the recall.

Precision is the fraction of retrieved instances that are relevant. Lesser the value of the precision means more the number of documents are relevant.

$$P = \frac{R_{Doc} - REL_{Doc}}{R_{Doc}}$$ (6)

Recall is the fraction of relevant instances that are retrieved.

$$R = \frac{R_{Doc} - REL_{Doc}}{REL_{Doc}}$$ (7)

F-measure is the weighted harmonic mean of precision and recall.

$$F-Measure = \frac{2 \times P \times R}{P + R}$$ (8)

E. Evaluation Methodology

Our system will generate an automatic level of feedback through which we evaluates the performance of the government recovery effort towards the damage/loss occurred during the disaster from dataset related to disaster and if the same data set is provided to the Under Graduate students, Post Graduate students and some other audience to manually annotated the level of performance of the government towards the
disaster. After matching the level of feedback performance of government towards the disaster from the manually calculated data and from the automatic feedback system and if the matching performance of our system is greater than 80% then our automatic feedback system is working well.

F. Baseline

As there is no prior work for exactly the same problem so we prepare our own baseline which uses Sentiment polarity score of words available in the people’s feedback

G. Experiment Results:

Table 1, 2, 3 shows the experimental evaluation results for damage/loss, government effort and people feedback for each disaster respectively. Here we use dataset1, dataset2, data set3 for the documents containing the news regarding damage/loss, government effort, and people feedback respectively for each disaster. Total score for each data set is calculated by summing up the \( Score_{imp} \) for each matching word with a vocabulary within the data set. After performing normalization upon the total scores of all the three data set for a particular disaster we get the evaluation score for each dataset for a particular disaster. Table 4 shows the scores of all the three disaster related events with the Comp_score and then comparing the Comp_score with the threshold table described in Table 5 to determine the performance of government during the disaster. Table 6 shows the evaluation metric table for the devised system. We calculated the evaluation metric for our devised system as well as for the people feedback which acts as a baseline for our devised system in which we used the dataset that we have collected out of which 50 documents are relevant to people feedback for Sikkim earthquake disaster and 60 documents are relevant to people feedback for Alberta flood disaster and 55 documents are relevant to Phailin cyclone. The precision of our devised system should be less than the baseline system.

<table>
<thead>
<tr>
<th>Disaster</th>
<th>Damage/Loss</th>
<th>Score (L)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sikkim Earthquake</td>
<td></td>
<td>0.86</td>
</tr>
<tr>
<td>Alberta Flood</td>
<td></td>
<td>0.89</td>
</tr>
<tr>
<td>Phailin Cyclone</td>
<td></td>
<td>0.97</td>
</tr>
</tbody>
</table>

Table 2. Experimental Evaluation of Government effort

<table>
<thead>
<tr>
<th>Disaster</th>
<th>Government Effort (G)</th>
<th>Score(G)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sikkim Earthquake</td>
<td>Data Set 2</td>
<td>0.95</td>
</tr>
<tr>
<td>Alberta Flood</td>
<td>Data Set 2</td>
<td>0.75</td>
</tr>
<tr>
<td>Phailin Cyclone</td>
<td>Data Set 2</td>
<td>0.91</td>
</tr>
</tbody>
</table>

Table 3. Experimental Evaluation of People feedback

<table>
<thead>
<tr>
<th>Disaster</th>
<th>People feedback (F)</th>
<th>Score(F)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Sikkim Earthquake</td>
<td>Data Set 3</td>
<td>0.54</td>
</tr>
<tr>
<td>Alberta Flood</td>
<td>Data Set 3</td>
<td>0.68</td>
</tr>
<tr>
<td>Phailin Cyclone</td>
<td>Data Set 3</td>
<td>0.56</td>
</tr>
</tbody>
</table>

Table 4. Disaster related events Scores

<table>
<thead>
<tr>
<th>Disaster</th>
<th>Score (L)</th>
<th>Score (G)</th>
<th>Score (F)</th>
<th>Comp_Score (A-B)</th>
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</thead>
<tbody>
<tr>
<td>Sikkim Earthquake</td>
<td>0.86</td>
<td>0.95</td>
<td>0.54</td>
<td>0.41</td>
</tr>
<tr>
<td>Alberta Flood</td>
<td>0.89</td>
<td>0.75</td>
<td>0.68</td>
<td>0.07</td>
</tr>
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<td>Phailin Cyclone</td>
<td>0.97</td>
<td>0.91</td>
<td>0.56</td>
<td>0.35</td>
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</table>

Table 5. Threshold based Risk Table

<table>
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<th>Threshold Range</th>
<th>Risk Value</th>
<th>Type of Govt.Effort</th>
</tr>
</thead>
<tbody>
<tr>
<td>P&lt;0.2</td>
<td>-2</td>
<td>Very Poor</td>
</tr>
<tr>
<td>0.2&lt;=P&lt;0.4</td>
<td>-1</td>
<td>Poor</td>
</tr>
<tr>
<td>0.4&lt;=P&lt;0.6</td>
<td>0</td>
<td>Average</td>
</tr>
<tr>
<td>0.6&lt;=P&lt;0.8</td>
<td>+1</td>
<td>Good</td>
</tr>
<tr>
<td>P&gt;0.8</td>
<td>+2</td>
<td>Very Good</td>
</tr>
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Table 6. Evaluation metric table

<table>
<thead>
<tr>
<th>Disaster</th>
<th>Evaluation Matrix</th>
<th>Devised System</th>
<th>Base Line</th>
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<tbody>
<tr>
<td>Sikkim Earthquake</td>
<td>Precision</td>
<td>Recall</td>
<td>F-Measure</td>
</tr>
<tr>
<td>Alberta Flood</td>
<td>Precision</td>
<td>Recall</td>
<td>F-Measure</td>
</tr>
<tr>
<td>Phailin Cyclone</td>
<td>Precision</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
V. CONCLUSION AND FUTURE WORK

In this paper, we develop an event linking algorithm using sentiment analysis based approach that will take the input in the form of documents containing news regarding damage/loss, recovery type news, news regarding people feedback and generates the output with the set of interlinked events between damage/loss, recovery type events and the people’s feedback during the disaster. Based upon the comparison between the people feedback and the threshold value we can generate quick response that depicts the performance of the government recovery effort with respect to the damage/loss occurred during the disaster. In the future this system can be used for reviewing the quality of a product or for movie reviewing.

REFERENCES


Location Based Search Privacy
Mechanism for Mobile Social Network

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Abstract— Mobile Social networking applications have become a very popular for communication and interaction, and participation of user has growing tremendously. There are many problems in mobile social networking sites. In this paper, we have proposed a location based search model for easily finding the new friends whose are matched with our interest. According to this model, we can interact with new environment and join our favorite activity online and also interact with that activity members, also provide the present location of users.

Keywords— Social network, Mobile computing, Mobile privacy, Location based search model.

I Introduction

In social networking sites, people can easily find and make new friends with other people with similar interests. Mobile social networking (MSN) services are used to communication with their friends. In this, members share their information with their mobile devices, and information may be experiences, interest, opinion and data. Mobile devices have the new capabilities as location related services. Which are as follows.

1) Users share their ideas in mobile social networking sites by the help of smart phone.

2) The main focus of mobile social networks is on mobile use like mobile communication, location-based services.

Mobile social networking integrates these two fast-growing services together. People walk around with their mobile devices and meet different people, known and unknown ones, every day, design systems for users to meet potential friends with similar interests or some other criteria. When two mobile devices are physically located close, they could start to exchange information without human interaction.

At the same time mobile phones are becoming more powerful and increasingly offer high speed Internet connectivity. Because of this people expect these social networking services to be available on their mobile device Mobile systems research is often done using ordinary smart phones. All major smart phone platforms, Android [9], BlackBerry [10], iPhone [11], Symbian [12] and Windows Mobile [13], support development of third party applications[13]. Each platform provides its own approaches to application development and application level resource management.

II Background

In this section we provide a short introduction to work in the area of mobile social networking and the technologies that have made it possible. The main parts of MSNs are following.

A. Mobile Computing

Mobile computing is called the portable and small computers, which have Personal Digital Assistants (PDA) like as mobile phones, laptops and palmtops etc. In this growing technological world, People are habitual to work o computer and internet. These two has become the most important part of life. Today every people want mobile devices because of their features and it works like a computer. It has also kept the data like a computer.
B. Social Networks
A social networking service is an online service, platform, or site that focuses on facilitating the building of social networks or social relations among people who, for example, share interests, activities, backgrounds, or real-life connections. A social network service consists of a representation of each user (often a profile), his/her social links, and a variety of additional services. Most social network services are web-based and provide means for users to interact over the Internet, such as e-mail and instant messaging. Online community services are sometimes considered as a social network service, though in a broader sense, social network service usually means an individual-centred service whereas online community services are group-centred. Social networking sites allow users to share ideas, activities, events, and interests within their individual networks.

Social networking is the grouping of individuals into specific groups, like small rural communities or a neighbourhood subdivision, if you will. Although social networking is possible in person, especially in the workplace, universities, and high schools, it is most popular online. This is because unlike most high schools, colleges, or workplaces, the internet is filled with millions of individuals who are looking to meet other people, to gather and share first-hand information and experiences about cooking, golfing, gardening, developing friendships, professional alliances, finding employment, business-to-business marketing and even groups sharing information about baking cookies to the Thrive Movement.

C. Existing Mobile Social Network Applications
MSN users [7] constantly search for ways to interact, engage, and share information while on the move through mobile devices (such as smart phones and tablets). Some newer devices support fourth-generation communication technologies, motivating vendors to provide services on a range of platforms, including Android, BlackBerry, OS, and Windows 8. In addition to hardware improvement, application developers are moving toward mobile advertising, TV, and social gaming, as well as toward new services (such as mobile wallets), mobile commerce, and cloud-based services. These services are enticing research topics. Many researchers have tackled these issues.

III Related Work
In [1] mobile users may face the risks of leaking their personal information and their location privacy. Propose is profile matching protocol, which enables one party to match its interest with the profile of another, without revealing its real interest and profile and vice versa. Authors have used blind transformation algorithm for easily finding the match profile. They have secured from many attacks as profile Leaking Attack, Runaway Attack.

In [3] have different privacy issues on online social networking sites and they said that when user create their profile once friends a small link connect their profile by photo, video, messenger and comment with each user profile by editing comments and sending messages. Authors have focused on the negative aspects of online social networking sites and control of personal information because when user place their information on public domain user can easily lose control over the data who sees if and who may use it.

In [2] explored the dual nature of friendship relations as an enabler but also as a pitfall for privacy in social networks. Authors have motivated the need for both public and private friend relationships in social networks and explain why maintaining public and private friend relationships in a centralized architecture is easier than in a peer-to-peer one. Friendship is the most fundamental relation in a social network. It is a relation between two members of the network and carries the understanding of friendship from the world. In a social network with a centralised authority and
clients being constantly able to establish a connection to this authority, hidden friendships can easily be implemented by views on a user's list of friends. A customised view on the outgoing friendship relations of a SN member is generated for each request. Our approach of securely hashing identifiers for hidden friendship relationships presents several theoretical and technical advantages which are particularly valuable for deployment in mobile social networking:

In [4] has discussed about the real life security issues and throats with facebook. They have discussed different type of facebook security issues as privacy and confidentiality, authentication and identity theft, intellectual property theft vandalism, harassment & stalking, data motion &disparagement, spam and cyber squatting. There all risks are greatest for facebook because fact is that people trust their facebook friends means that identity theft is greater.

In [5] has proposed Find U, a set of privacy-preserving profile matching schemes for proximity-based mobile social networks. In Find U, an initiating user can find from a group of users the one whose profile best matches with his/her. And has also proposed novel protocols that realize each of the user privacy levels, which can also be personalized by the users. Author has defined problem of the system model, Adversary Model, Design Goals, then solve the problem. Find the friend to use same protocol and User discovery the friend and establishment Key to easily find friends.

In [6] has described the Mobile Online Social Networks (MOSN) in our study exhibit some leakage of private information to third parties. In this paper presented a taxonomy of ways to study privacy leakage and report on the current status of known leakages A device such as a smart phone could be used to access the mobile or full web site via a mobile browser as well as a device-specific application tailored for a MOSN. They have to be aware of the duration of any privacy setting they have made. When they allow some information, such as location, to be used by the MOSN for a legitimate purpose, they have to be aware that it might be handed over to third-parties. They have used the following criteria of inclusion of candidate MOSNs for our study.

IV Security Problems
The general problems of securities are discussed as follows.

A. Mobile Privacy Issues
Smart phones and other mobiles are just like mini computers. They all have the power and functionality of computers. Our mobiles contain large amount of personal information like contact numbers, photos, videos. Security or privacy [8] risks are inherent to the internet. Our personal information are become the target of malware and spyware. Mobile devices contain the user information that are not usually found on personal computer as telephone calls, text messages, and history of our location.

Other challenge is the devices, small screens, which makes the effective communication. But users have no many options for privacy. Today the application economy is thriving. Mobile app is a development stage, in which developers are focusing on getting new products to market quickly as possible. Many mobile applications didn’t provide the privacy of users. This represents not just a failure, but it is also suggest a lack of attention to application privacy.

V Location Based Search and Privacy Solution
We have tried to provide the solution by location based search model. This is the searching process, through this users will able to search their interested group activities like cricket, news, football and all that in a unknown environment.
Fig.1. Activity search model

In Figure 1. Admin creates group activities, and there different- different admin of different- different place. For example: A user goes to different place and wants to join his interested activity like cricket. Then he will search that activity, user connects to the admin through the location server. Because location server connects this user to the new environment and show user’s interested activity nearby of that environment. After getting his interested group activity, user will automatically connect to admin of that group. But they both are not able to permit to see all information of each other. Admin will see only user’s name and profession. And user will see only related information of group activities. User will not interact to admin’s profile.

There is no direct interaction between user and admin, they both are connect through the activity groups. According to this model, we can easily interact with a new environment.

The procedure for proposed work is as follows.
1. Register.
2. Login.
3. Server automatically search the current location of users.
4. Users search of their interested activities like cricket, gym, hospital, yoga so on. nearest to the current location.
5. Activities are list out in front of user.
6. Users select their interested activity.
7. Send the request to that group (request goes to admin of that activity).
8. Admin verify the details of users like name, age, date of birth and profession only. After that accept or reject.
9. If accept, users add in group.
10. Users can able to see the details about the group and also see group members.
11. Users are permitted only for seeing the group members but not for seeing the profile of any member of that group.

Fig.2. Location based search model

This model is a process of system. In figure 2. admin creates a account and location of admin is held on server. The all information related to the admin is kept on database. Location on the server of the admin will be changed according to the admin. And then a user is also create a account and admin both are log in their account. Admin upload a file on his account to multiple user. Admin also generate the public and private key. If user is on a new place than server will search current location of user and user wants to join activity group of his nearby. User
searches his interest group and easily find the group. If admin of that area has already created that group. After that user will send the friend request to admin to joining that group. Than admin will check the user’s profile than accept otherwise reject. If admin accept the request of user download the file from admin’s profile with private key but related to that activity. And if admin rejects the request of user. According to this model user can easily find the new friends whose are matched with user’s interest. This will be used especially for saving the time and through this user can easily interact with new environment. After meetings, users will increase their social network. In this main thing is that user only goes inside the group activity not in admin’s profile. Information of admin will be kept safely on the database.

VI Conclusion

We have discussed a solution for easily finding the friends with our interest in the shortest time of duration. In this paper, we have proposed a location based search model for saving our time. Through this, we can direct interact with new friends, and create the connection between new friends. Main thing is that we would have great friend circle. After that, users will able to easily find out their interested activities and create a trusted social network. This paper is implementing in Java Scripting Programming language.

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A Novel method for Image Segmentation & Comparative Study

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Abstract – Multimedia, Search engines, Digital libraries, Medical and Geographic Databases and Criminal investigation are using Image Databases in several distinct application domains. The evolution of techniques for acquisition, transmission and storage of images has also allowed the construction of very large image databases. All these factors have spurred great interest in content-based image retrieval (CBIR) techniques. Existing CBIR systems based on low-level features (such as color and texture) can be classified into three main categories: (1) global approaches, (2) partition based approaches and (3) regional approaches. The objective of this paper is to present the image segmentation process for image. Segmentation is a challenging problem, and its accuracy implies more in the system construction. Its main purpose is to filter the image with median filter, mean filter and fused with global thresholding method to produce better segmented results. This is more reliable segmentation during hand shape change and hand crossing. The approach is demonstrated with comparison of conventional methods of filters and segmentation process. Three segmentation methods are experimented over here. This greatly facilitates the image localization process which is important for the purpose of recognition.

Keywords: CBIR, Thresholding, Edge detection,

1. INTRODUCTION

Image segmentation can be defined as a process that partitions a digital image into disjoint (non-overlapping) regions. A region is a connected set of pixels—that is, a set in which all pixels are adjacent or touching. Thus in a connected set, a connected path can be traced between any two pixels without ever leaving the set [58]. When a human observer views a scene, the processing that takes place in the visual system essentially segments the scene for him or her. This is done so effectively that one sees not a complex scene. With digital processing, however, the objects in an image are laboriously isolated by breaking up the image into sets of pixels, each of which is the image of one object. Image segmentation is typically used to locate objects and boundaries (lines, curves, etc.) in images [45]. More precisely, image segmentation is the process of assigning a label to every pixel in an image such that pixels with the same label share certain visual characteristics. The result of image segmentation is a set of segments that collectively cover the entire image, or a set of contours extracted from the image. Each of the pixels in a region is similar with respect to some characteristic or computed property, such as color, intensity, or texture. Adjacent regions are significantly different with respect to the same characteristic [60]. When applied to a stack of images, typical in Medical imaging, the resulting contours after image segmentation can be used to create 3D reconstructions with the help of interpolation algorithms like marching cubes.

Figure 1.1 Basic Segmentation Process [71]

Several general-purpose algorithms and techniques have been developed for image segmentation. Since there is no general solution to the image segmentation problem, these techniques often have to be combined with domain knowledge in order to effectively solve an image segmentation problem for a problem domain [50].

1.1 Different Types of Image Segmentation:

1.1.1 Thresholding
The simplest method of image segmentation is called the thresholding method. This method is based on a clip-level (or a threshold value) to turn a gray-scale image into a binary image. The key of this method is to select the threshold value (or values when multiple-levels are selected) [39]. Several popular methods are used in industry including the maximum entropy method, Otsu's method (maximum variance), and et al. k-means clustering can also be used.

1.1.2 Clustering methods
The K-means algorithm is an iterative technique that is used to partition an image into K clusters. The basic algorithm is:
1. Pick K cluster centers, either randomly or based on some heuristic.
2. Assign each pixel in the image to the cluster that minimizes the distance between the pixel and the cluster center.
3. Re-compute the cluster centers by averaging all of the pixels in the cluster.
4. Repeat steps 2 and 3 until convergence is attained (e.g. no pixels change clusters)

1.1.3 Compression-based methods
Compression based methods postulate that the optimal segmentation is the one that minimizes, over all possible segmentations, the coding length of the data. The connection between these two concepts is that segmentation tries to find patterns in an image and any regularity in the image can be used to compress it. The method describes each segment by its texture and boundary shape [16]. Each of these components is modeled by a probability distribution function and its coding length is computed as follows:

1. The boundary encoding leverages the fact that regions in natural images tend to have a smooth contour. This prior is used by Huffman coding to encode the difference chain code of the contours in an image. Thus, the smoother a boundary is, the shorter the coding length it attains.
2. Texture is encoded by lossy compression in a way similar to minimum description length (MDL) principle, but here the length of the data given the model is approximated by the number of samples times the entropy of the model. The texture in each region is modeled by a multivariate normal distribution whose entropy has closed form expression [22]. An interesting property of this model is that the estimated entropy bounds the true entropy of the data from above. This is because among all distributions with a given mean and covariance, normal distribution has the largest entropy. Thus, the true coding length cannot be more than what the algorithm tries to minimize.

1.1.4 Histogram-based methods
Histogram-based methods are very efficient when compared to other image segmentation methods because they typically require only one pass through the pixels. In this technique, a histogram is computed from all of the pixels in the image, and the peaks and valleys in the histogram are used to locate the clusters in the image [42]. One disadvantage of the histogram-seeking method is that it may be difficult to identify significant peaks and valleys in the image. In this technique of image classification distance metric and integrated region matching are familiar.

1.1.5 Edge detection
Edge detection is a well-developed field on its own within image processing. Region boundaries and edges are closely related, since there is often a sharp adjustment in intensity at the region boundaries. Edge detection techniques have therefore been used as the base of another segmentation technique [43]. The edges identified by edge detection are often disconnected. To segment an object from an image however, one needs closed region boundaries. Edge is nothing but boundary between two images. Edge detection technique refers to the identification and locating the sharp discontinuities in the image.

1.1.6 Region growing methods
The first region growing method was the seeded region growing method. This method takes a set of seeds as input along with the image. The seeds mark each of the objects to be segmented. The regions are iteratively grown by comparing all unallocated neighboring pixels to the regions. The difference between a pixel's intensity value and the region's mean, δ, is used as a measure of similarity [55]. The pixel with the smallest difference measured this way is allocated to the respective region. This process continues until all pixels are allocated to a region. Seeded region growing requires seeds as additional input. The segmentation results are dependent on the choice of seeds. Noise in the image can cause the seeds to be poorly placed. Unseeded region growing is a modified algorithm that doesn't require explicit seeds. It starts off with a single region A1 – the pixel chosen here does not significantly influence final
segmentation. At each iteration, it considers the neighboring pixels in the same way as seeded region growing. It differs from seeded region growing in that if the minimum δ is less than a predefined threshold T then it is added to the respective region Aj. If not, then the pixel is considered significantly different from all current regions Ai and a new region An + 1 is created with this pixel.

1.1.7 Graph partitioning methods
Graph partitioning methods can effectively be used for image segmentation. In these methods, the image is modeled as a weighted, undirected graph. Usually a pixel or a group of pixels are associated with nodes and edge weights define the (dis)similarity between the neighborhood pixels. The graph (image) is then partitioned according to a criterion designed to model "good" clusters [29]. Each partition of the nodes (pixels) output from these algorithms are considered an object segment in the image. Some popular algorithms of this category are normalized cuts, random walker minimum cut, and isoperimetric partitioning and minimum spanning tree-based segmentation.

1.1.8 Model based segmentation
The central assumption of such an approach is that structures of interest/organs have a repetitive form of geometry. Therefore, one can seek for a probabilistic model towards explaining the variation of the shape of the organ and then when segmenting an image impose constraints using this model as prior. Such a task involves (i) registration of the training examples to a common pose, (ii) probabilistic representation of the variation of the registered samples, and (iii) statistical inference between the model and the image [11]. State of the art methods in the literature for knowledge-based segmentation involve active shape and appearance models, active contours and deformable templates and level-set based methods.

1.1.9 Multi-scale segmentation
Image segmentations are computed at multiple scales in scale-space and sometimes propagated from coarse to fine scales; Segmentation criteria can be arbitrarily complex and may take into account global as well as local criteria. A common requirement is that each region must be connected in some sense.

2 BACKGROUND
Among the several challenges faced by the image processing community, segmentation and localization of homogeneous object regions in an image remains a daunting proposition. This is primarily due to the complex nature of the data handled therein. In the binary domain, the task of segmentation reduces to the problem of background subtraction. It substitutes the object centric features with an intensity level complementary to the background perspective. The end result is an aggregation of extracted object regions out of the image background. The problem of segmentation becomes more severe for multilevel and color images. This is mainly due to the variety of the gray scale and color intensity gamut [56].

2.1.1 Histogram Shape-Based methods, where, for example, the peaks, valleys and curvatures of the smoothed histogram are analyzed. This category of methods achieves Thresholding based on shape properties of the histogram. The image histogram acts as a graphical representation of tonal distribution in a digital image. It plots the number of pixels for each tonal value. By looking at the histogram for a specific image a viewer will be able to judge the entire tonal distribution at a glance [23]. Histograms are constructed by splitting the range of the data into equal-sized bins (called classes). Then for each bin, the number of points from the data set that fall into each bin is counted. Vertical axis represents: Frequency (i.e., counts for each bin) or say the number of pixels in particular tone and Horizontal axis represents: Response variable/tonal variations.

The shape property comes into play in different forms- convex hull Thresholding or two peaks representation. Convex hull Thresholding: Rosenfeld’s method is based on analyzing the concavities of histogram h(g) vis- a- vis its convex hull , Hull H(g), that is the set theoretic difference[Hull H(g) – h(g)]. When the convex hull of the histogram is calculated, the deepest concavity point becomes candidate for a Thresholding. In competing concavities, some object attribute feedback, such as low busyness of threshold image edges could be used to select one of them [22]. Peak and Valley Thresholding: The Peak & Valley provides a way to locate peaks (maximum points) and valleys (minimum points) along a specified line in an image. This is similar to the Edge Probe but instead of detecting large changes in intensity this module will detect the change in direction of
intensity. This is useful in detecting patterns, frequencies or reoccurring objects.

Figure 2.1 Image of a Finger Print & Histogram of finger print [72].
The left side of the horizontal axis represents the black and dark areas, the middle represents medium grey and the right side represents light and pure white areas. The vertical axis represents the size of the area that is captured in each one of these tones [25].
In a dark image, the grey levels (histogram) would be clustered at the lower end. In a uniformly bright image, the grey levels would be clustered at the upper end. In a well contrasted image, the grey levels would be well spread out over much of the image [25].

2.1.2 Clustering-Based Methods, where the gray-level samples are clustered in two parts as background and foreground (object), or alternately are modeled as a mixture of two Gaussians. Clustering Thresholding suggests minimizing the weighted sum of within class variance of the foreground and background pixels to establish an optimal threshold. Recall that Minimization of within class variance is tantamount to the maximization of between-class scatter. This method gives satisfactory results when the numbers of pixels in each class are close to each other [40].

2.1.3 Entropy-Based methods result in algorithms that use the entropy of the foreground and background regions, the cross-entropy between the original and binaries image, etc. This class of algorithms exploits the entropy of the distribution of the gray levels in a scene. The maximization of the entropy of the threshold image is interpreted as indicative of maximum information transfer. Other authors try to minimize the cross-entropy between the input gray-level image and the output binary image as indicative of preservation of information. Johannsen and Bille and Pal, King, Hashim were the first to study Shannon entropy based Thresholding. Entropic Thresholding of Kapur - In this method the foreground and background classes are considered as two different sources. When the sum of the two class entropies is a maximum the image is said to be optimally threshold [2]. Yen, Chang and Chang have considered a multilevel Thresholding scheme where in addition to the class entropies a cost function based on the number of bits needed to represent the threshold image is included. Cross-entropic Thresholding of Li - In this method the threshold determination is formulated as a constrained maximum entropy inference problem. The constraint forces the total intensity in the reconstructed image to be identical to that in the observed image in both the foreground and background regions.

2.1.4 Object Attribute-Based Methods search a measure of similarity between the gray-level and the binarized images, such as fuzzy shape similarity, edge coincidence, etc. The algorithms considered under this category select the threshold value based on some similarity measure between the original image and the binarized version of the image. These attributes can take the form of edges, shapes, or one can directly consider the original gray-level image to binary image resemblance. Alternately they consider certain image attributes such as compactness or connectivity of the objects resulting from the binarization process or the coincidence of the edge fields [9].

2.1.5 Spatial Methods that use higher-order probability distribution and/or correlation between pixels. In this class of algorithms one utilizes spatial information of object and background pixels, for example, in the form of context probabilities, correlation functions, co-occurrence probabilities, local linear dependence models of pixels, two-dimensional entropy etc. One of the first to explore spatial information was Rosenfeld who considered such ideas as local average gray level for thresholding [45]. Other authors have used relaxation to improve on the binary map as in, the Laplacian of the images to enhance histograms, the quad tree thresholding, and second-order statistics. Co-occurrence probabilities have been used as indicator of spatial dependence as in Lie, Pal, and Chang. Recently Leung and Lam have considered thresholding in the context of a posteriori spatial probability estimation [21].

Several methods have been proposed to binarize an 3 PROPOSED IMAGE SEGMENTATION TECHNIQUE.
The input of the system can be colored image or grey image, if the image is grey then directly compute its pixel value and label the pixels accordingly otherwise first convert the colored image in to grey image and then repeat the procedure. After labeling apply the segmentation technique, in our proposed work we apply Thresholding method to segment an image. The output of the system is a segmented image, which can be useful or can be taken as input in many systems such as face recognition system, finger print matching etc. The resultant segmented images are reduced in size require less storage space and have the relevant information as well. The procedure stops when classification of all pixels is done as an object or background. The object’s pixels can be grouped and can be used further.

3.1 Different Approaches/Methods for image Segmentation

Image compression works by reducing the amount of visible information. So whereas the original picture might be able to display 65 million colors, since the eye can not differentiate between anywhere near this number of colors the software is going through and removing colors that are not needed. The actual complexity of the algorithms is far more complex than that, and beyond the scope of this response. But the threshold has to do with how the algorithm decides what information to throw away and what information to keep, and thus will affect the resulting quality and size of the image after compressions [10]. The proposed work will be dependent on the Threshold segmentation in which an image is break into blocks/segments by assigning the threshold value \([T]\) and finding an appropriate value of \(T\) which would result in an better image then source image in terms of simplicity or smoothness.

3.2 Approaches to Image Segmentation:

3.2.1 Thresholding: The simplest method of image segmentation is called the thresholding method. This method is based on a clip-level (or a threshold value) to turn a gray-scale image into a binary image. The key of this method is to select the threshold value. Thresholding based image segmentation aims to partition an input image into pixels of two or more values through comparison of pixel values with the predefined threshold value \(T\) individually.

3.2.1.1 Method:

During the thresholding process, individual pixels in an image are marked as "object" pixels if their value is greater than some threshold value (assuming an object to be brighter than the background) and as "background" pixels otherwise.
3.2.1.2 Threshold selection
The key parameter in the thresholding process is the choice of the threshold value (or values, as mentioned earlier). Several different methods for choosing a threshold exist; users can manually choose a threshold value, or a thresholding algorithm can compute a value automatically, which is known as automatic thresholding [61].

A threshold image $g(x, y)$ is defined as

$$g(x, y) = \begin{cases} 
1 & \text{if } f(x, y) > T \\
0 & \text{if } f(x, y) \leq T 
\end{cases}$$

When $T$ depends only on $f(x, y)$ the threshold is called Global, if $T$ depends on both $f(x, y)$ (gray level of a point) and $p(x, y)$ (property of the point) the threshold is called a Local and if depend on the spatial coordinates $X$ and $Y$ then called as Adaptive threshold.

3.2.1.3 Global Thresholding
In the simplest implementation of boundary location by thresholding, the value of the threshold Gray level is held constant throughout the image. If the background gray level is reasonably constant throughout, and if the objects all have approximately equal contrast above the background, then a fixed global threshold will usually work well, provided that the threshold gray level is properly selected. Segmentation is then accomplished by scanning the image pixel by pixel and labeling each pixel as object or background, depending on whether the gray level of that pixel is greater or less than the value of $T$.

$$g(x, y) = \begin{cases} 
1 & \text{if } f(x, y) > T \\
0 & \text{if } f(x, y) \leq T 
\end{cases}$$

3.2.2 Edge-based
Edge based segmentation is the location of pixels in the image that correspond to the boundaries of the objects seen in the image. It is then assumed that since it is a boundary of a region or an object then it is closed and that the number of objects of interest is equal to the number of boundaries in an image.

3.2.3 Region based
The region based segmentation is partitioning of an image into similar/homogenous areas of connected pixels through the application of homogeneity/similarity criteria among candidate sets of pixels. Each of the pixels in a region is similar with respect to some characteristics or computed property such as color, intensity and/or texture. The assumption in these techniques is that the partitions that are formed correspond to objects or meaningful parts of the image.

3.3 Focused Techniques for Image Segmentation:
The proposed work is mainly focused on Thresholding method.

Figure 3.2.a) original image [74]
Figure 3.2.b) Global threshold [74]

Figure 3.2.a shows the original image of rice grains and Figure 3.2.b shows the resultant image by applying global threshold on original image.
First initially select the value on which segmentation is done. Here are the 3 measures for this.

### 3.3.1 Random selection
In this method, a random value of threshold is selected and on the basis of the selected value segmentation is done. The pixels having value less than the selected value of \( t \) is treated as the background pixel and the pixels having greater value than \( t \) is treated as object pixel. The value of \( t \) can be low, high or somewhere between that. If the value is very low then the resultant image is darker and if it is chosen high then the resultant image is whiter. If the value of \( t \) is choose properly, this is the best method for segmentation. For this method the colored image is first converted in to gray image and then segmentation is done.

### 3.3.2 Mean based
This is a low pass filter, which removes high spatial frequencies from an image and is also good at reducing Gaussian noise present in an image. It works by passing a mask (usually 3x3) over the image calculating the mean intensity for the mask and setting the central pixel to this value. To calculate the mean each value in the mask will be multiplied with it's respective value in the block and the results summed to give the mean intensity for the block. If we were to run this filter over an entire image it would have the effect of smoothing out the image.

“Take an average across element's neighborhood”.

The formula is simple — sum up elements and divide the sum by the number of elements.

![Figure 3.7 taking an average](image)

### 3.3.3 Median based
Median filtering is one kind of smoothing technique. The median filter is a nonlinear digital filtering technique, often used to remove noise. Median filtering is very widely used in digital image processing because, under certain conditions, it preserves edges while removing noise. It also does a better job than the mean filter at preserving edges within an image. The main idea of the median filter is to run through the signal entry by entry, replacing each entry with the median of neighboring entries. The pattern of neighbors is called the "window", which slides, entry by entry, over the entire signal. The median is calculated by first sorting all the pixel values from the surrounding neighborhood into numerical order and then replacing the pixel being considered with the middle pixel value. (If the neighborhood under consideration contains an even number of pixels, the average of the two middle pixel values is used).

The median filter, when applied to grayscale images, is a neighborhood brightness-ranking algorithm that works by first placing the brightness values of the pixels from each neighborhood in ascending order. The median or middle value of this ordered sequence is then selected as the representative brightness value for that neighborhood. Subsequently, each pixel of the filtered image is defined as the median brightness value of its corresponding neighborhood in the original image. The median filter for color images operates differently from the grayscale median filter. Since each pixel in an RGB color image is composed of three components (red, green, and blue), it is not useful to rank the pixels in the neighborhood according to brightness. Instead, the color median filter works by comparing each pixel's color to that of every other pixel in the neighborhood. The pixel whose red, green, and blue components have the smallest sum of squared differences from the color coordinates of its neighbors is then chosen to replace the central pixel of the neighborhood.

### 4 Conclusion and Future Scope
Till date various techniques for image segmentation had been developed. But none of the technique developed so far optimistic for image segmentation. Each segmentation technique have their respective cons such as Edge- based have problem of noisy images and smooth boundaries and Region- based takes high computational time which forces a requirement to have an optimistic technique for image segmentation. The objective of the proposed work is try to find an optimistic approach for image segmentation for the purpose of reducing the network load while transferring an image or object extraction. The main focus of work is on thresholding technique and to find an appropriate value for it by applying random selection, mean and median. The random selection is quite simple and easy to implement but there is a of decision making. Median provides good result as compared to mean.
and removes noise by making the image smoother but does not provide much detail about the image. If the threshold value is choose wisely then this is the best method for segmentation. It provide better result if the value of t is chosen somewhere between 60 to 200

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Notes by Shi and Malik on Segmentation, computer vision, March 2004.


Performance Comparison of Instrumentation Amplifiers – A Beginner’s View

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Abstract—Instrumentation amplifiers (IAs) have become inevitable in all kinds of sensitive instrumentation and control applications. For an Engineer, a brief and qualitative idea of how an instrumentation amplifier works will be useful in a multitude of ways. In this paper, we describe and suggest three different configurations and modifications that can be used to design an Instrumentation Amplifier – Difference Amplifier, 3 Stage IA, IA with linear gain - for various applications. Main theme of the paper is to describe the flaws and advantages of using these configurations with proper justification. Effort has been made to bring out various obvious implementation issues which greatly affect the final design as per applications.

Keywords — Instrumentation; Amplifier; linear gain; CMRR; Temperature dependence.

INTRODUCTION

Signals produced by transducers are very small levels of currents or voltages, often indistinguishable from noise. Instrumentation Amplifiers (IAs) are the back bones in Process control and Instrumentation applications, which help in sensing the quantity to be measured and bring them to a manageable level for further processing and controlling. Their usage as transducer signal conditioning circuits puts them often in harsh and rapidly changing environments with varying noise and interference from unwanted sources. So an IA must have the ability to amplify the desired signal and reject unwanted signals, intelligently in a sense. In this respect, an IA must have the following characteristics [1]:

- Ability to separate signals from ambient noise.
- Ability to amplify the signal as per a desired linear transfer characteristic.
- Ability to sense signals from sensor and amplify it without loading it.
- Insensitivity to ambient environmental changes and interferences.

With these considerations in mind, we proceed to describe the three basic configurations – Difference Amplifier (DA), 3 - stage Instrumentation Amplifier and Linear gain IA. Main aim of the paper is to highlight the following aspects of Instrumentation Amplifier design:

- A beginner has some trouble designing an IA. It is the case that most of the times, bread board implementations do not lead to the desired results unless some definite procedures are followed. Effort will be made to make fundamental concepts clear.
- Effect of temperature on the overall performance of the IA will be elaborated with support from simulation results.
- Resistance mismatches in the circuits commonly used cause unforeseen variations to the output. Such effects have been simulated and definite conclusions will be drawn from the results.

We present different simulation results making some conclusions obvious regarding choosing a configuration. All these simulations have been done using NI Multisim® software and graphs plotted using MATLAB®.

PRELIMINARIES

Some introduction to the terminology associated with signal conditioning and very much related to an IA design is necessary to be highlighted, so as to make the rest of the paper easy to understand. These
terms are the ones used by manufacturers as well, in their data sheets [1].

- Ratio of Output voltage to the input signal voltage when the input applied is not common to the input terminals of an IA is called **Differential Gain** ($A_d$). If the applied input is common to both the terminals, the ratio is called **Common mode Gain** ($A_{cm}$).

- Ratio of Differential mode gain to common mode gain is called **Common Mode Rejection Ratio** (CMRR). It is expressed in decibels as,

$$\text{CMRR (dB)} = 20 \log_{10} \left( \frac{A_d}{A_{cm}} \right). \quad (1)$$

Other terms used in literature will be defined when they are encountered later in the paper. The three terms defined above provide very efficient metrics to qualify an IA.

**THE DIFFERENCE AMPLIFIER**

The basic building block which gives an IA its ability to reject common mode noise is the Difference Amplifier. For applications which do not require high accuracy, a DA alone can do the job with a very simple circuit configuration, using a single operational amplifier (Op – Amp). The circuit is shown in Fig. 1. In a DA, we give one signal to the inverting terminal, to get an output given by

$$V_{\text{out}(i)} = - \left( \frac{R_2}{R_1} \right) V_{\text{in}}, \quad (2)$$

and the other signal to the non-inverting terminal, because of which the output becomes

$$V_{\text{out}(ni)} = \left[ \frac{R_4}{(R_3 + R_4)} \right] (1 + \frac{R_2}{R_1}) V_{\text{in}}. \quad (3)$$

Fig. 1 The Difference Amplifier (Gain = 100)

When we use the principle of superposition, the final output due to both the sources becomes

$$V_{\text{out}} = V_{\text{out}(i)} + V_{\text{out}(ni)} . \quad (4)$$

Now, when the resistances are randomly selected, two signals are not amplified by equal amounts. This is catastrophic if the DA has to operate in the presence of Common mode noise. Hence, to avoid this problem, we impose the following condition [1][2],

$$(R_2/R_1) = (R_4/R_3), \quad (5)$$

which reduces the output equation to simply

$$V_{\text{out}} = (R_2/R_1) (V_{\text{NI}} - V_I) \quad (6)$$

Where $V_I$ is the input given to inverting terminal and $V_{\text{NI}}$ is that given to non-inverting terminal, both referenced to a single reference voltage, for example, ground voltage. This specification of reference is important because the difference between the signals is going to be amplified here and this depends on the reference potential considered.

**Some Inherent Problematic Issues**

Though the DA seems deceptively simple, even the slightest of the changes can ruin its desired functionality. The following are some of the factors that influence the output of the DA, as observed through simulations [5].

- The circuit provides appreciable common mode gain, which is because of different input impedances offered by the inverting and non-inverting terminals.
- Also, the circuit is unstable when temperature and tolerance changes are encountered during circuit operation, as seen from the AC analysis.
- It is observed that as long as the bridge is balanced, gain at room temperature remains fairly constant.
- At very low voltages, peak in gain is observed, because of the offset voltage, still being amplified by the circuit. Since the circuit is solely producing gain, it makes offset errors much difficult to deal with.

But, it is easy to design, less space consuming and low power consuming because of just one Op - Amp. The disadvantages being that it provides low CMRR and less stability with temperature and at high frequencies. Gain variation is very difficult to achieve because any variation in resistances will disturb the bridge balance and deteriorates the output. The solution comes from a simple modification of the circuit which becomes quite obvious as we go on. The modified circuit tends to be much better than the basic difference amplifier, making it the most popular configuration used in industry. The circuit is 3-Op Amp.
Instrumentation Amplifier, which has become synonymous with the name Instrumentation Amplifier.

**Op–Amp Instrumentation Amplifier**

The circuit schematic diagram of the 3–Op Amp Instrumentation Amplifier is shown in Fig. 2. A similar analysis done in the case of DA will help in getting the equations for designing this IA [2]. For the difference amplifier stage, we have again the condition,

\[ \frac{R_2}{R_1} = \frac{R_4}{R_3} \]  

(7)

And when \( R_5 = R_6 \), the equation for output voltage becomes [2]

\[ V_{out} = [1 + (2\times R_3 / R_g)] \times \left( R_2 / R_1 \right) \times (V_{in2} - V_{in1}) \]  

(8)

Now, going into the functioning, the circuit has two stages: The input Buffering Stage and the Difference Amplifier Stage.

**Input Buffering Stage:** It helps in
- Providing high input impedance for both inverting and non-inverting terminals, to avoid loading.
- Providing equal input impedances for both terminals to provide symmetry.
- Increasing CMRR by doing buffering.
- By avoiding offset errors of the Op Amps used in the input buffering stage by using dual Op Amp ICs in which Op Amps track each other closely and provides the gain stage as well.

Differential input signals are amplified by the Equation (8) while common mode signals are rejected here itself. Common Mode signals are just buffered with unity gain by each of the two buffering Op Amps. At the resistance \( R_g \), these voltages are subtracted. Difference Amplifier’s role is clear by now. It just makes the subtraction and removes the noise. Gain is not provided by this stage. Offset error of input buffer stage are compensated by using closely tracking Op Amps. Offset errors of difference amplifier stage can be minimized using the right combination of resistors. This stage does not add any gain, so errors can be set to a minimum always, since we are not touching this section even in case of gain variation requirements.

Disadvantages include issues namely, the number of Op – Amps required, leading to more power consumption and area. Gain variation is possible by \( R_g \), but it is not linear. To make the gain linear, another variation the design is used widely, as discussed in the next section.

**Linear Gain Instrumentation Amplifier**

The circuit schematic for this design is shown in Fig. 3. It is seen that addition of one more op amp in the feedback path of Difference Amplifier stage causes some effect in the operation.

![Fig. 2 3 Op–Amp Instrumentation Amplifier](image)

**Fig. 2 3 Op–Amp Instrumentation Amplifier**

And when \( R_5 = R_6 \), the equation for output voltage becomes [2]

![Fig. 3 Instrumentation Amplifier with Linear Gain](image)

**Fig. 3 Instrumentation Amplifier with Linear Gain**

The offset error of the additional op amp is responsible for some reduction in CMRR. When compared with previous design, this design provides linear gain variation, but also reduces the noise rejection capability. Linear gain variation with simple adjustment of \( R_g \) is the only additional
advantage this circuit provides over the generic 3-stage Op Amp 1A configuration. Using the analysis shown in [2], we have, under bridge balance condition, and \( R_5 = R_6 \),

\[
V_{out} = \left[1 + \left(2 \frac{R_5}{R_3}\right)\right]\left[\left(\frac{R_2}{R_1}\right) \times \left(\frac{R_g}{R_f}\right)\right] (V_{in2} - V_{in1})
\]

(9)

Apart providing the advantage of making gain linear with \( R_g \), this circuit is disadvantageous in the sense that it needs additional Op – Amp for proper functionality, increasing the area and power requirements. The following sections show the simulation results and conclusions are drawn on the basis of the results in the graphs.

**SIMULATIONS**

Before going to simulations, it must be noted that the Op – Amps used are the generic 741 models available in Multisim®. The model parameters are as listed here in Table I. It has to be noted that offset compensation is not done in these cases, to see the effect of such unwanted voltages on the outputs.

<table>
<thead>
<tr>
<th>S. No.</th>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Input Resistance</td>
<td>2 MΩ</td>
</tr>
<tr>
<td>2</td>
<td>Open Loop Gain</td>
<td>200000</td>
</tr>
<tr>
<td>3</td>
<td>Output Resistance</td>
<td>75 Ω</td>
</tr>
<tr>
<td>4</td>
<td>CMRR</td>
<td>90 dB</td>
</tr>
<tr>
<td>5</td>
<td>Input Offset Voltage (V os)</td>
<td>1 mV</td>
</tr>
<tr>
<td>6</td>
<td>Input Offset Current (I os)</td>
<td>20 nA</td>
</tr>
<tr>
<td>7</td>
<td>Input Bias Current (I bs)</td>
<td>80 nA</td>
</tr>
</tbody>
</table>

**TABLE I : Model Parameters for 741 Op – Amp in simulations**

Difference Amplifier (DA)

From Fig. 4, it is clear that component mismatches and variations over large ranges in DA can cause dramatic changes in gain non – linearly. Both common and differential mode gains follow a similar pattern but, common mode output voltage for the designed circuit is 11.2 while that of differential mode is 100. This makes CMRR equal to 19.015 dB. It is obvious that the performance is poor. To improve in this aspect, the following steps are suggested:

- Maintaining \( R_1 = R_3 ; R_2 = R_4 \) will remove the offset voltage caused by Input Bias currents.
- Internally compensated Op – Amps are to be used to remove the effect of Offset Voltages. If not possible, external offset compensation is to be adopted.
- Components used must be very precise and sufficiently of high values.

![Fig. 4 DA – Variation of Common and Differential mode gains with changes in resistor R2.](image)

Because, at high resistance values, slight variations and mismatches in their values do not affect the output much. If low value resistors are used, slight changes in their resistances, even changes in the lengths of connecting wires will give rise to some common mode output which may not be acceptable to the operation.

Fig. 5, shows the variation of gain with temperature for a DA. All resistor values shown in the circuit of Fig. 1, are at a temperature of 27 °C. It is clear that the differential gain does not change much and hence is stable for the given resistance temperature coefficients considered. But, the common mode gain changes more. Even a change of temperature by one degree can cause common mode gain to change by 3.5 %. To solve this problem to some extent, the following steps are suggested:

- Resistors of very low temperature coefficients of resistance which are less vulnerable to temperature fluctuations are to be used.
- Precision Op – Amps with temperature insensitive characteristics are to be used.
- If both such cases are not possible then resistors can be replaced with potentiometers and frequent
calibration is needed to remove the common mode noise. This is not practical in many situations. As far as AC analysis is concerned, the circuit follows a predictable variation and not much variation in terms of bandwidth. The next section is about the 3 Op – Amp IA.

**3 Op – Amp Instrumentation Amplifier**

Fig. 6 shows the variation of gain of the IA shown in Fig. 2. It must be noted that the difference amplifier stage of this circuit is still subject to all the effects discussed in the previous sections. To minimize those effects in this case, we choose the gain of the difference amplifier stage to be unity. Some straightforward conclusions can be drawn from the comparison of gain curves for both DA and 3 Op – Amp IA. They are summarized below.

- It is evident that the gain can be varied in a predictable but non-linear fashion with variation in \( R_g \). With such variations, the common mode gain does not change. Even the offsets of the buffer Op – Amps appear as common mode voltage to the difference amplifier stage and they are rejected. Further external compensation and using dual Op – Amps for the input stages are good design methodologies to bring down the common mode gain further.
- The CMRR in this case if proportional to differential gain, which is desired characteristic. Changing \( R_g \) can take CMRR to as high as 90 or 100 dBs. \( R_g \) here is the gain setting element.
- A microcontroller or an analog linearization circuit can be used to make the gain variation linear, which is desirable.

Now, going to the temperature dependencies, we have the curve of Fig. 7. The following conclusions can be made out.

- Common mode gain changes here by 0.7% for one degree change in temperature for the same difference amplifier stage design. So, temperature stability is obviously better in this case.
- It is observed that common and differential mode gains are almost equally changing and hence, effect on CMRR is very less with temperature. In fact, at the given normal temperature of 27 °C, CMRR of the circuit is 33.13 dB, for the given value of \( R_g \) equal to 1 KΩ. Further reduction of \( R_g \) will cause better CMRR.
compared to the DA alone. Even in the case of AC operation, the phase and gain plots show a non-linear variation with increasing $R_g$ and a linear variation with increase in temperature. With temperature increase, the bandwidth gets slightly reduced, similar to the DA case.

3 Op – Amp Instrumentation Amplifier with linear gain

Fig. 8, shows the effect of changing $R_g$ in the feedback element of the circuit shown in Fig. 3. This is the gain variation element. From the graph, it is observed that

- Both common and differential mode gains are linear varying in nature with changing $R_g$.
- But, CMRR here also changes because common mode noise is not constant but it also varies. The reason for this is that the Op – Amp sitting in the feedback part of the circuit injects some voltage of its own and hence spoils the CMRR. To avoid this, the feedback Op –Amp must be an ultra-low input current, compensated Op –Amp.
- All the design methodologies followed in the previous sections must be followed here as well. But, it has to be observed that the feedback given here at the difference amplifier stage if positive. Because, the Op – Amp in the feedback network will invert the feedback signal automatically. In effect, we eventually end up obtaining a negative feedback.

It has to be noted that the same linear gain variant can be implemented with just a DA as well. The connection of feedback Op –Amp remains the same in both cases. In DA implementation, using this will improve the performance slightly.

In this circuit implementation, there is a slight variation in AC response as well. At low values of gain, an additional pole is obtained in the response. At higher gains, this poles effect reduces and the response tends to be that which can be observed in the previous two cases. This additional effect is because of the inclusion of the Op –Amp in the feedback path. Care should be taken when high frequency applications demand the usage of this kind of circuit configuration.

COMPARISONS AND CONCLUSIONS

Table II, gives the comparative results of all the three configurations considered in this paper for Instrumentation Amplification applications. It is clear from the above table that the 3 Op –Amp IA has the best stability with temperature. DA has the least. Linear gain variant has an intermediate nature of stability.

<table>
<thead>
<tr>
<th>TABLE II : Comparison of Gain Variations</th>
<th>Fig. 11 IA with linear gain – Variation of common mode and differential mode gains with temperature</th>
</tr>
</thead>
</table>

Fig. 10 IA with linear gain – Variation of Common and Differential mode gains with $R_g$
Other variations like the 2 Op – Amp IAs and linear circuits with T bridges are also possible to be implemented. The final decision is left to the designer on the basis of the application in hand and balancing the requirements while compensating for the losses and non-idealities.

**REFERENCES**


A Weighted Relevance Response Approach for Web Image Retrieval Using Textual and Visual Features

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Abstract
In this paper, we propose a unified relevance feedback framework for Web image retrieval. Our framework shows advantage over traditional RF mechanisms in the following three aspects. First, during the RF process, both textual feature and visual feature are used in a sequential way. To seamlessly combine textual feature-based RF and visual feature-based RF, a query concept-dependent fusion strategy is automatically learned. Second, the textual feature-based RF mechanism employs an effective search result clustering (SRC) algorithm to obtain salient phrases, based on which we could construct an accurate and low-dimensional textual space for the resulting Web images. Thus, we could integrate RF into Web image retrieval in a practical way. Last, a new user interface (UI) is proposed to support implicit RF. On the one hand, unlike traditional RF UI which enforces users to make explicit judgment on the results, the new UI regards the users’ click-through data as implicit relevance feedback in order to release burden from the users.

I. INTRODUCTION
A common limitation of most of the existing Web image retrieval systems is that their search process is passive, i.e., disregarding the informative interactions between users and retrieval systems. An active system should get the user into the loop so that personalized results could be provided for the specific user. To be active, the system could take advantage of relevance feedback techniques. Relevance feedback, originally developed for information retrieval in a practical way, an efficient and effective mechanism is required for constructing an accurate and low-dimensional textual space with respect to the resulting Web images.

Although all existing commercial Web image retrieval systems solely depend on textual information, Web images are characterized by both textual and visual features. Image retrieval has the following two characteristics when comparing with text retrieval. In this paper, we propose a unified relevance feedback framework for Web image retrieval. There are three main contributions of the paper.

In this paper, we propose a unified relevance feedback framework for Web image retrieval. There are three main contributions of the paper.

- A dynamic multimodal fusion scheme is proposed to seamlessly combine textual feature-based RF (TBRF) and visual feature-based RF (VBRF).
- More specifically, a TBRF algorithm is first used to quickly select a possibly relevant image set. Then, a VBRF algorithm is combined with the TBRF algorithm to further re-rank the resulting Web images.
- The fusion of VBRF and TBRF is query concept dependent and automatically learned. The textual feature-based RF mechanism employs an effective search result clustering (SRC) algorithm.
- A new UI is proposed to support implicit RF.

II. DYNAMIC MULTIMODAL FUSION
A. Image Representation
The images collected from several photo forum sites, e.g., photosig [9], have rich metadata such as title, category, photographer’s comment and other people’s...
critiques. These images constitute the evaluation dataset for the proposed relevance feedback framework. For example, a photo of photosig1 has the following metadata. In order to facilitate later citation of this photo, we denote it by

- **Title**: *early morning*.
- **Category**: *landscape, nature, rural*.
- **Comment**: *I found this special light one early morning in Pyrenees along the Vicdessos river near our house*.

- **One of the critiques**: *wow I like this picture very much I guess the light has to do with everything the light is great on the snow and on the sky (strange looking sky by the way) greatly composed nice crafted border a beauty.*

To represent the textual feature, vector space model [10] with TF-IDF weighting scheme is adopted. More specifically, the textual feature of an image is an dimensional vector and can be given by

\[ F^T = (\omega_1, \ldots, \omega_L) \]

where

- \( N \) is the total number of images;
- \( N_i \) is the number of images whose metadata contains the \( i \)th term
- \( t_f_i \) is the frequency of \( i \)th term in \( I \)'s textual space

\[ \omega_i = \frac{N}{\log \frac{N}{n_i}} \]

\[ t_f_i = \frac{N_D}{D(w)} \]

\[ D(w) = \sum_{D(w)} \cos(d_i, c) \]

\[ e = \frac{1}{|D(w)|} \sum_{D(w)} d_i \]

\[ CE = -\sum_i \frac{D(w) \cap D(t)}{|D(w)|} \log \frac{D(w) \cap D(t)}{D(t)} \]

\[ IND = \frac{IND_t + IND_r}{2} \]

\[ IND_t = -\sum_i \frac{f(i)}{TF} \log \frac{f(i)}{TF} \]

\[ F_{opt} = F_{ini} + \alpha \sum_{r \in Rel} F_r - \beta \sum_{j \in Non-Rel} F_j \]

\( L \) is the number of all distinct terms of all images’ textual space

\( \omega_i \) is the weight of \( i \)th term in \( I \)'s textual space

**B. RF in Textual Space**

To perform RF in textual space, Rocchio’s algorithm [4-6] is used. The algorithm was developed in the mid-1960s and has been proven to be one of the most effective RF algorithms in information retrieval. The key idea of Rocchio’s algorithm is to construct a so-called optimal query so that the difference between the average score of a relevant document and the average score of a nonrelevant document is maximized. Cosine similarity is used to calculate the similarity between an image and the optimal query. Since only clicked images are available for our proposed framework, we assume clicked images to be relevant and define the feature of optimal query as follows:

\[ TFIDF = f(w) \cdot \log \frac{N}{D(w)} \]  \hspace{1cm} (1.9)

\[ LEN = n \]  \hspace{1cm} (1.10)

\[ ICS = \frac{1}{|D(w)|} \sum_{d_i \in D(w)} \cos(d_i, c) \]  \hspace{1cm} (1.11)

\[ e = \frac{1}{|D(w)|} \sum_{d_i \in D(w)} d_i \]  \hspace{1cm} (1.12)

\[ CE = -\sum_i \frac{D(w) \cap D(t)}{|D(w)|} \log \frac{D(w) \cap D(t)}{D(t)} \]  \hspace{1cm} (1.13)

\[ IND = \frac{IND_t + IND_r}{2} \]  \hspace{1cm} (1.14)

\[ IND_t = -\sum_i \frac{f(i)}{TF} \log \frac{f(i)}{TF} \]  \hspace{1cm} (1.15)

\[ F_{opt} = F_{ini} + \alpha \sum_{r \in Rel} F_r - \beta \sum_{j \in Non-Rel} F_j \]  \hspace{1cm} (1.16)
C. RF in Visual Space
To perform RF in visual space, Rui’s algorithm [15] is used. Assume clicked images to be relevant, both an optimal query and feature weights are learned from the clicked images. More specifically, the feature vector of the optimal query is the mean of all features of clicked images. The weight of a feature dimension is proportional to the inverse of the standard deviation of the feature values of all clicked images [15]. Weighted Euclidean distance is used to calculate the distance between an image and the optimal query. Although Rui’s algorithm is used currently, any RF algorithm using only relevant images could be used in the unified framework.

D. Dynamic Multimodal Fusion
There has been some work on fusion of relevance feedback in different feature spaces [16]–[18]. A straightforward and widely used strategy is linear combination [16], [17]. Nonlinear combination using support vector machine (SVM) was proposed in [18]. Since the super-kernel fusion algorithm [18] needs irrelevant images, it is incapable for systems only offering relevant images. Since textual features are more semantic-oriented and efficient than visual features while visual features have finer descriptive granularity than textual features, we combine the RF in both feature spaces in a sequential way. The flowchart of the RF of our unified framework is shown in Fig. 1. First, RF in textual space is performed to rank the initial resulting images using the optimal query learned in (1.3). Then, RF in visual space is performed to re-rank the top images. The re-ranking process is based on a dynamic linear combination of the RF in both visual and textual spaces. Note that restricting the re-ranking only on the top images has two advantages. First, the relevance of the top images could be guaranteed by the former RF in textual space. Second, the efficiency of RF process could be ensured, for RF in visual space could possibly be inefficient on a very large image set.
III. SRC-BASED TEXTUAL SPACE CONSTRUCTION

To construct an accurate and low-dimensional textual space for the resulting Web images, we use the SRC algorithm proposed in [19]. The author re-formalizes the clustering problem as a salient phrase ranking problem.

\[
S = \beta \cdot S^V + (1 - \beta) S^T 
\]

\[
\beta = \alpha \cdot \exp(-\lambda \cdot D_{ave}) 
\]

\[
D_{ave} = \frac{1}{n} \sum_{i=1}^{n} ||F_{q}^i - F_{opt}^i||/n 
\]

\[
S^V = 1 - D^V 
\]

\[
S^T = \frac{1}{n} \sum_{i=1}^{n} F_{opt}^i 
\]

where:

• \( S \) is the similarity metric in both visual and textual spaces;
• \( S^V \) is the similarity between \( I \)'s visual feature and \( F_{opt}^i \);
• \( S^T \) is the cosine similarity between \( I \)'s textual feature and \( F_{opt}^i \);
• \( \beta \) is the dynamic linear combination parameter for similarity metric in both visual and textual spaces;
• \( \alpha \) and \( \lambda \) are parameters which control the relative contribution of RF in visual space;
• \( D_{ave} \) is the deviation of the clicked image in visual space

Given the above five properties, we use a single formula to combine them and calculate a single salience score for each phrase. In our case, each term \( x \) can be a vector learned from previous training data is then applied to combine \( x = (TFIDF, LEN, ICS, CE, IND) \) the five properties into a single salience score \( Y \), the performance of linear regression is the best one.

\[
y = b_0 + \sum_{j=1}^{p} b_j x_j + e 
\]

based on which we use the (1.1) and (1.2) to compute the textual feature.
IV. FRIENDLY USER INTERFACE

![Flowchart of the UI](image)

V. EXPERIMENTAL RESULTS

A. Evaluation Dataset
To construct the evaluation dataset, approximately three million images were crawled from several photo forum sites, e.g., photosig [9]. To automatically evaluate our proposed SRC-based RF mechanism, an image subset was selected and manually labeled as follows. First, ten representative queries were chosen. Then, for each query, the key terms related to the

B. Evaluation of RF Fusion
The proposed RF fusion strategy (TVRF) has three parameters that need to be determined. $\alpha$ controls the overall contribution of RF in visual space. $\lambda$ fine tunes the contribution, and the scope $K$ in which the resulting images are re-ranked by the combination of the textual similarities and the visual similarities. Because $K$ is less correlated with $\alpha$ and $\lambda$, we first chose based on a simplified version of (1.4) by constraining $\lambda$ to 0, i.e.,

$$S= \alpha \cdot S^V + (1- \alpha) S^T.$$

Four RF strategies were evaluated and compared: RF using textual feature only (TBRF),
E. Evaluation of SRC-Based RF
In our experiment, two RF strategies were evaluated and compared: traditional RF and the proposed SRC-based RF. Both of them use Rochhio’s algorithm to construct a so-called optimal query. The difference lies in constructing the textual space for the resulting images. Traditional RF uses all terms present in the metadata to construct the textual space, while the SRC-based RF uses the SRC algorithm to obtain the salient phrases, based on which the textual space is constructed.

D. Efficiency of TVRF
In order to evaluate the real time performance of the proposed technique, efficiency performance of the proposed RF fusion strategy (TVRF) is worth discussing as well. Since there are two textual-based RF mechanisms available in our work, we refer to the SRC-based TVRF as SRC-TVRF.

VI. CONCLUSION
In this paper, we have presented a unified relevance feedback framework for Web image retrieval. During RF process, both textual features and visual features are used in a sequential way. A dynamic multimodal fusion strategy is proposed to seamlessly combine the RF in textual space and that in visual space. To integrate RF into Web image retrieval in a practical way, the textual feature-based RF mechanism employs an effective search result clustering (SRC) algorithm to construct an accurate and low-dimensional textual space for the resulting Web images. Besides explicit relevance feedback, implicit relevance feedback, e.g., click-through data, can also be integrated into the proposed mechanism. Then, a new user interface (UI) is proposed to support implicit RF. Experimental results on a database consisting of nearly three million Web images show that the proposed mechanism is wieldy, scalable, and effective.

REFERENCES


An Adaptive Fuzzy Algorithm For Solving Economic Power Dispatch Problem Considering Uncertainties

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Abstract- This paper describes the design of an adaptive fuzzy algorithm for the solution of economic power dispatch problem. In the economic dispatch problem there is a station with NG generators committed and the active power load PD is given a real power generation PG, for each generator has to be allocated so as to minimize the total cost. The above constrained optimization problem is solved using genetic algorithm. This method employs coefficients of variation in the genetic algorithm to account for uncertainties.

Index terms- Fuzzy sets, Economic Dispatch problem, Genetic algorithm, Crossover, Mutation.

I. INTRODUCTION

Suppose there is a station with NG generators committed and the active power load PD is given, the real power generation PG, for each generator has to be allocated so as to minimize the total cost [1]. The optimization problem can therefore be stated as follows:

Minimize:

\[ F(Pgi) = \sum Fi (Pgi) \quad (i=1......NG) \]

Subject to

(i) the energy balance equation

\[ \sum Pgi = PD \quad (i=1......NG) \]

(ii) and the inequality constraints

\[ Pgi (\text{min}) < Pgi < Pgi (\text{max}) \]

\[ (i = 1,2,....... , NG) \]

where

\[ Pgi \] is the decision variable, i.e, power generation

\[ PD \] is the real power demand

\[ NG \] is the number of plants

\[ Pgi (\text{min}) \] is the lower permissible limit of real power generator

\[ Pgi (\text{max}) \] is the upper permissible limit of real power generator

\[ Fi(Pgi) \] is the operating fuel cost of the ith plant and is given by the quadratic equation.

\[ Fi(Pgi) = aiPgi^2 + biPgi +ci \quad \text{Rs/h} \]

II. AN OVERVIEW OF FUZZY LOGIC

Based on the nature of fuzzy human thinking Lotfi Zadeh, a computer scientist at the University of California, Berkeley, originated the "Fuzzy Logic" or Fuzzy set theory, in 1965.

In Fuzzy logic different types of membership functions are used viz. triangular membership function, trapezoidal, sigmoid-right etc.

The various operations that can be performed on fuzzy sets are listed below:

1. Union: \[ \max( \muA (x), \muB(x)) \] (This is equivalent to Boolean or)

2. Intersection: \[ \min( \muA (x), \muB(x)) \] (This is equivalent to Boolean And)

3. Complement or Negation: \[ 1- \muA (x) \] (This is equivalent to Boolean Not)

4. Product of two fuzzy sets: \[ \muA (x) \cdot \muB (x) \]

5. Multiplying fuzzy set by a crisp number: \[ k \cdot \muA (x) \]

6. Power of a fuzzy set: \[ [ \muA (x)]^m \] etc.

A fuzzy inference process consists of the following five steps:

Step 1: Fuzzification of input variables.

Step 2: Application of fuzzy operator (And, or, not) in the If (Antecedent) part of the rule.

Step 3: Implication from the antecedent to the consequent (Then part of the rule).
Step 4: Aggregation of the consequents across rules.

Step 5: Defuzzification.

The various implication methods used in fuzzy logic are listed below:

1. **Mamdani type:** The output membership value is union (or) of all the component membership functions.

2. **Lusing-Larson Type:** The output membership function is scaled instead of being truncated.

3. **Sugeno type or Takagi-Sugeno-Kang type (1985):** The output membership functions are only constants or have linear relationships with inputs.

The various defuzzification methods used are listed below:

1. **Centre of Area method:** The geometric centre of the output is taken. The output membership value is formed by taking union of all contributions of rules whose degree of fulfillment is greater than 0.

   \[ Z_0 = \frac{\sum Z_i \cdot \mu_{out}(Z_i)}{\sum \mu_{out}(Z_i)} \]

   \[ i=1 \ldots n \]

2. **Height method:** This method considers only the height of each contributing membership function at the mid-point of the base.

3. **Mean of maxima method:** In this method only the highest membership function component in the output is considered. If M such maxima are present then output is given by:

   \[ Z_0 = \frac{\sum Z_m}{M} \]

   \[ m=1 \ldots M \]

4. **Sugeno method:** In this method the output is determined as follows:

   **Zero-order method:**

   \[ Z_0 = (K_1 \cdot DOF_1 + K_2 \cdot DOF_2 + K_3 \cdot DOF_3) / (DOF_1 + DOF_2 + DOF_3) \]

   where \( DOF_i \) indicates the degree of fulfillment of \( i^{th} \) membership function.

   **First-order method:**

   \[ Z_0 = (Z_1 \cdot DOF_1 + Z_2 \cdot DOF_2 + Z_3 \cdot DOF_3) / (DOF_1 + DOF_2 + DOF_3) \]

**III. AN OVERVIEW OF GENETIC ALGORITHM**

A global optimization technique known as genetic algorithm (GA) has emerged as a candidate due to its flexibility and efficiency for many optimization applications [1]. Genetic algorithm is a stochastic searching algorithm.

The steps of the genetic algorithm are listed below:

**Step 1:** Code the problem variables into binary strings.

**Step 2:** Randomly generate initial population strings. Tossing of a coin can be used.

**Step 3:** Evaluate fitness values of population members.

**Step 4:** Is solution available among the population?

   If ‘yes’ then go to Step 9.

**Step 5:** Select highly fit strings as parents and produce offsprings according to their fitness.

**Step 6:** Create new strings by mating current offspring. Apply crossover and mutation operators to introduce variations and from new strings.

**Step 7:** New strings replace existing one.

**Step 8:** Go to Step 4 and repeat.

**Step 9:** Stop.

Implementation of a problem in a genetic algorithm starts from the parameter encoding. Binary coded strings having 0s and 1s are used.

Genetic algorithms mimic the survival-of-the-fittest principle of nature to make a search process. Fitness function \( f(x) \) is derived from the objective function and used in successive genetic operations.

In crossover operator, information is exchanged among strings of the mating pool to create new strings. The various crossover operators used are:
one point crossover, multipoint crossover, uniform crossover etc.

Mutation is another important operator used in genetic algorithm. Mutation operator changes 1 to 0 and vice versa with a small mutation probability.

IV. SOLUTION OF ECONOMIC DISPATCH PROBLEM USING GENETIC ALGORITHM:

The solution of Economic dispatch problem using genetic algorithm is given in the following steps:

Step 1: Declare parameters used in the algorithm:

Fitness_value[21],
selected_count,
selected_count2,
selected_cross,
sel[21],
Output_string[21],
cross_site,
child_string[21],
lm_data[21]

Step2: Declare global functions:

Wheel_selection()

Step3: Read parameters:

The values of a1, a2, a3, b1, b2, b3, c1, c2, c3, p1min,
p1max, p2min, p2max,
P3min, p3max and
B[i][j] parameters

Step4: Initialize the values of parameters:

Length = 16
L = 20
Pc = 0.8 probability of crossover
Pm = 0.01 probability of mutation
Lm_min = 10.0
Lm_max = 12.5

Step5: while

(population_counter <= 20)
{
    For ( i=0 to 16)
    {
        Generate a random number
        If (random number <= probability of flip)
            String[i] = 1;
        Else
            String[i] = 0;
    }
    // end of for loop
    Y=0;
    For (i=0 to 16)
        Y=y+(string[i]*2

Lm=lm_min +
((( lm_max - lm_min)/
(2^length-1))*y);

Lm_data[population_counter]=lm;

New_factor = 2*a2;
// coefficient of variation =
// standard deviation / mean =σ/mean
Cvp1=0.25;
Cvp2=0.5;
CF1=(a1 + new_factor)*cvp1*cvp2;
Changed_factor1 = CF1;
CF2=(a2 + new_factor) *cvp1*cvp2;
Changed_factor2 = CF2;
CF3=(a3 + new_factor) *cvp1*cvp2;
\[ Changed\_factor3 = CF3; \]

// solve simultaneous equations
\[ A_{ii} = 2 \times (a_i + \text{changed\_factor}_i + (1m \times B_{ii})); \]
\[ A_{ij} = 2 \times lm \times B_{ij}; \]
\[ A_{ji} = A_{ij} \]

SEC_{i} = lm - b_{i}; \quad i = 1, 2, 3.

// solve the equations
\[ A_{11} \times p_{1} + A_{12} \times p_{2} + A_{13} \times p_{3} = SEC_{1} \]
\[ A_{21} \times p_{1} + A_{22} \times p_{2} + A_{23} \times p_{3} = SEC_{2} \]
\[ A_{31} \times p_{1} + A_{32} \times p_{2} + A_{33} \times p_{3} = SEC_{3} \]

// check for limits
If (p_{i} < p_{\text{imin}})
\[ P_{i} = p_{\text{imin}}; \]
Else
\[ If (p_{i} > p_{\text{imax}}) \]
\[ P_{i} = p_{\text{imax}}; \quad i = 1, 2, 3 \]

// calculate total _ cost
\[ \text{Total\_cost} = \sum a_{i} \times p_{i}^{2} + b_{i} \times p_{i} + c_{i}; \quad i = 1…3 \]

// calculate ploss
\[ \text{Ploss} = \sum \sum p_{i} \times B[i][k] \times p_{k} \]
\[ (i = 1…3, j = 1…3) \]
\[ E_{p} = \text{fabs}(pd + ploss - (p_{1} + p_{2} + p_{3})); \]
( Note: The function fabs is used to find absolute value of float data type )

(pd = specified load demand)

// Calculate fitness
\[ F = \frac{1}{1 + \alpha \times E_{p} / pd}; \]
assuming \( \alpha = 1.0 \)

Fitness\_value[\text{population\_count}]=f;

// increment population\_counter by 1
Population\_counter = population\_counter + 1;

} // End of while loop

Step 6: Call the function wheel\_selection() to make a table of selected string using Wheel Selection algorithm.

Step 7: Perform crossover operation.

Step 8: Perform mutation operation.

Step 9: Execute steps 1 to 8 using Generation 2. The steps 1 to 8 can be executed on N Generations till required accuracy of ep and lm is achieved.

The above algorithm takes into consideration the coefficients of variation cvp1 and cvp2[3]. These parameters modify the algorithm to take into consideration the uncertainties.

The output when these parameters are zero is given in Table 1.

Table 2 contains the output fitness and lm when cvp1 and cvp2 are non-zero

**TABLE 1:**

<table>
<thead>
<tr>
<th>Population</th>
<th>lm</th>
<th>f</th>
<th>selected string</th>
<th>binary string</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>(lagrangian multiplier) (fitness)</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>1</td>
<td>11.428283</td>
<td>0.943670</td>
<td>1</td>
<td>10000010010001001</td>
</tr>
<tr>
<td>2</td>
<td>11.288739</td>
<td>0.894882</td>
<td>1</td>
<td>11101111111000001</td>
</tr>
</tbody>
</table>
maximum fitness= 0.959617 and lm= 11.678912 */

-------------------------------------------------------------------------------------------------------------------------------

TABLE 2: /* The parameters: New factor= 0.01218 , cvp1=0.25 , cvp2=0.5

-------------------------------------------------------------------------------------------------------------------------------

Population     lm  f       selected string  binary string
(lagrangian multiplier) (fitness)  
-------------------------------------------------------------------------------------------------------------------------------
1  11.428283  0.805211  1  1000001001001001
2  11.288739  0.775401  2  1110111110000001
3  10.449492  0.633537  3  1110000001110101
4  12.431067  0.922490  4  0000111110011111
5  12.232815  0.978609  4  1100001010011001
6  11.810559  0.897504  5  0100111110001111
7  10.600214  0.719358  5  0110111101111000
8  11.23064  0.842369  6  0000000011001110
9  11.678912  0.959617  6  1101011111010101
10 11.743992  0.936237  7  1010100101001101
11 12.339818  0.769750  8  0011011111111011
12 12.274586  0.784660  8  0101011100010111
13 11.419509  0.940467  9  1101101010001001
14 10.624208  0.724012  9  1101011111111100
15 11.741512  0.937104  10 0010101001001101
16 12.446403  0.756029  10 0100000101011111
17 10.858244  0.772452  11 0100011111110100
18 11.313268  0.903137  12 010111001100001
19 10.973335  0.799284  13 1101010111000110
20 11.360151  0.919288  13 1110001011010001
TABLE 2:

<p>| | | | | |</p>
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>9</td>
<td>11.678912</td>
<td>0.863796</td>
<td>6</td>
<td>1101011111010101</td>
</tr>
<tr>
<td>10</td>
<td>11.743992</td>
<td>0.880187</td>
<td>7</td>
<td>1010100101001101</td>
</tr>
<tr>
<td>11</td>
<td>12.339818</td>
<td>0.947380</td>
<td>8</td>
<td>0001100111110111</td>
</tr>
<tr>
<td>12</td>
<td>12.274586</td>
<td>0.966142</td>
<td>9</td>
<td>0101011100010111</td>
</tr>
</tbody>
</table>

Maximum fitness = 0.978609 and lm=12.232815 */

V. ALGORITHM FOR CROSSOVER OPERATOR USING FUZZY RULES:

The following steps are used for crossover operator[1]. This operator is modified using fuzzy rules.

Cvp1: coefficient1 of variation.
   [This is used with Penalty term]
Cvp2: coefficient2 of variation.

Step 1: Generate an integer number between 1 and 20.
   Index=generated integer
   Selected_count = sel[Index]
   The corresponding entry from sel array is selected.

Step 2: Generate second integer number between 1 and 19.
   Index=second generated integer
   Selected_count2=sel[Index]

Step 3: Fuzzy rules to identify the crossover site:

Rule 1: if cvp1 is LOW and cvp2 is HIGH
   Then selected_cross is between 0 to 9.

Rule 2: if cvp1 is MEDIUM and cvp2 is HIGH
   Then selected_cross is between 0 to 12.

Rule 3: if cvp1 is HIGH and cvp2 is (HIGH or MEDIUM)
   Then selected_cross is
Cvp1 is LOW if cvp1 < 0.1.
Cvp1 is MEDIUM if value lies between 0.1 and 0.25.
Cvp1 is HIGH if cvp1 > 0.25.
Cvp2 is LOW if cvp2 < 0.1.
Cvp2 is MEDIUM if value lies between 0.1 and 0.45.
Cvp2 is HIGH if cvp2 > 0.45.

VI. FUZZY RULES BASED MUTATION OPERATOR

The following steps are used for mutation operator [1]. The operator is modified using Fuzzy rules.

Cvp1: coefficient1 of variation. [This is used with Penalty term]
Cvp2: coefficient2 of variation.

Step1: Generate a random number between 0 and 1.
Assign the number to selected_string.
Step2. If cvp1 is LOW and cvp2 is LOW then mutation site is between 0 to 5.
Step3: If cvp1 is MEDIUM and cvp2 is LOW then mutation site is between 0 to 10.
Step4: If cvp1 is HIGH and cvp2 is HIGH then mutation site is between 0 to 15.

VII. CONCLUSION

The economic dispatch problem solution is tested using the genetic algorithm. The algorithm is modified to take into consideration the uncertainties. Fuzzy rules are devised to modify existing crossover and mutation operator used in genetic algorithm. These fuzzy rules are based on coefficient of variation; all these rules are tested depending upon the value of the coefficients. These are the new terms introduced in the algorithm which takes into consideration uncertainties.

REFERENCES

ABSTRACT Programmers often copy code fragments or other design documents and paste it with or without modification. These types of similar code fragments or design documents are known as software clone. Cloned code is considered harmful for various reasons. A clone contains multiple and unnecessary duplicate fragments of code. Due to clones maintenance, costs are increased. They make software products inconsistent as making changes to a cloned code can create faults and lead to unexpected behaviour. Similarly, Duplicated fragments and parts of models are also harmful in model-based development such as UML. Model

Keywords: UML Domain Models, Model Clones, Clone Detection, UML Parsing.

1. INTRODUCTION
Programmers often copy code fragments or other design documents and paste it with or without modification. These types of similar code fragments or design documents are known as software clone[1]. In previous reports it is found that the total amount of software cloning in software system ranges from 5-15% and can be even 50 % of code base. If any code fragment contains a bug and all the other fragments that are copied from it, may also suffer from the same problem. Thus clones are considered as “bad smell” [2] in software product and produce project maintenance difficult. So that clone evolution analysis has also become an important part of managing code clones.

There are various reasons of occurrence of clones in the software. Developers create the clones in sufficient time to create proper abstractions, to meet project deadlines because they may not have

same problem that encountered before.

Developers may also create clones sometimes to obtain maintenance such as to avoid risk in developing new code.

Code Clones
According to the definition of similarity, if two code fragments that are similar to each other, are called clones to each other. The similarity between the fragments can be defined by textual representation or on their functionality representation code. Clones can be defined further in four categories.

Source code clones are copies or near-copies of other portions of code, often created by copying and pasting portions of source code. This working session is concerned with building a communal research infrastructure for clone detection. The intention of this working session is to try to build a consensus on how to continue to build a benchmark suite and results archive for clone and source comparison-related research and development. The working session is structured to foster discussion and debates over what should be collected in the archive, and how to make it best to improve clone detection research techniques.

The copying of code has been studied within software engineering mostly in the area of clone analysis. Software clones are regions of source code which are highly similar; these regions of similarity are called clones, clone classes, or clone pairs [1].

While there are several reasons, why two regions of code may be similar, the majority of the clone analysis literature attributes cloning activity to the intentional copying and duplication of code by
Programmers; clones may also be attributable to automatically generated code, or the constraints imposed by the use of a particular framework or library [3].

Cloning is the unnecessary duplication of data whether it is at design level or at coding level. It results to excessive maintenance costs as well. So cut paste programming form of software reuse deceivingly raise the number of lines of code without expected reduction in maintenance costs associated with other forms of reuse.

The reasons, why programmers duplicate codes include the following reasons [4]:

Making a copy of a code fragment is simpler and faster than writing the code from scratch. Also, the fragment may already be tested so the introduction of a bug seems less likely.

Evaluating the performance of a programmer depends on the amount of code produces by programmer gives a natural incentive for copying code.

Efficiency considerations may make the cost of a procedure call or method invocation seems too high a price. In industrial software development contexts, time pressure together with first and second points lead to plenty of opportunities for code duplication [4].

1.2. Model Clones

Models are duplicate fragments of Architecture of Model of the project. It is difficult to formulate the actual definition of a model clone because of the abstract nature of a model. It can, however define a model clone as a set of similar or identical fragments in a model.

According to the definition of cloning, there can be different notions of similarity. Clones can be based on text, lexical or syntactic structure or can be semantics, model based functionally. Clones can even be similar if it follows the same pattern, that is, the same building plan.

This phenonmenon occurs similarly in models, suggesting that model clones are as detrimental to model quality as they are to code quality. However, programming language code and visual models have significant differences that make it difficult to directly transfer notions and algorithms developed in the code clone arena to model clones. The structural clone analysis extends the benefits of analysis based on simple clones in the areas of program understanding, maintenance, reuse, and refactoring.

Reference Model

Looking at the Package Diagrams, it can be easily seen that both the Models are abstraction of ATM banking model, which may or may not be clones. The reference Model (Package) has two internal packages defined in the main Package called “Banking”.

One Package is ATM and another package is...
Accounts. The ATM Internal Package shows relationship/Dependencies with Account Package. Both Packages have different-2 classes. The Classes of ATM Internal Package also shows relationship with Classes of Account Package.

On the other hand, there are two occurrences of Classes named “ATM”, so “ATM” is definitely a clone candidate.

Possible Model Clone (Candidate)

Fig 3. A Possible Model Clone in UML 2.0 Package Diagrams

It is still not obvious, though which reference in the diagram refers to which model element, or whether they refer to the same model element or not. To external viewers, who are often equating models and diagrams, this model element will be invisible. Four major challenges regarding model clones are:

3. To understand the structure of real clones and derive a practical definition of model clones.
4. To quantitatively analyze the structure of medium to large scale models and develop method to detect clones.
5. To derive a formal framework for model clones and develop an algorithm detects clones in models of realistic size and structure.
6. To implement the algorithm and method, balancing precision and recall against acceptable run time.

Why is Cloning used in Software Systems?

In general, software systems contain a significant amount of cloned code and the amount of cloning varies depending on the domain and origin of software systems. There are various factors for which clones can be introduced in a system. Developers may create clones to meet project deadlines because they may not have enough time to create proper abstractions or the short-term cost of creating those abstractions may outweigh the benefits of creating the duplicated code. Developers may also create clones to solve the same problems encountered before but reluctant to do abstraction because one do not understand the solution, do not have enough time or simply because one do not care about the impact of making clones. Toomim et al identified a set of cases that makes the abstraction costly and leads programmers to leave the cloned code instead. Kapser et al identified a set of eight cloning patterns that explain the motivation of cloning, and also listed their advantages and disadvantages. A comprehensive list of factors that introduce cloning can be found in the survey by Roy and Cordy. This list categorized the reasons for cloning in the following four groups.

2. RELATED WORK

Storrle Harald [1], describes that, Code clones, have been identified as major source of software quality issues. Evidence suggests that this phenomenon occurs similarly in models, suggesting that model clones are as detrimental to model quality as they are to code quality. The Author proposes a formal definition of model clones, specify a clone detection algorithm for UML domain models, and implement it prototypically. The problem with code clones is of course that they are linked only by their similarity, i.e., implicitly rather than explicitly which makes it difficult to detect them. The paper also discusses that the clones are a substantial problem for code based development, and model clones are increasingly becoming a problem for model-based development.

Papers & Researches carried out in this field, are described/detailed below:

JClone: Syntax tree based clone detection for Java (April 2010)
This Work examines a software engineering process to create an abstract syntax tree based clone detector for the projects implemented in Java programming language. An unavoidable amount of money is spent on maintaining existing software systems today. Software maintenance cost generally higher than development cost of the system, therefore, lowering maintenance cost is highly appreciated in software industry.

Detecting duplicate code fragments can significantly decrease the time and effort, therefore, the maintenance cost. Clone code detection can be achieved via analyzing the source code of given software system. An abstract syntax tree based clone detector for Java systems is designed and implemented through this study.

Towards clone detection in UML domain models (Dec. 2010)

Model clone is a set of similar or identical fragments in a model. Clones are a substantial problem for code based development, and model clones are increasingly becoming a problem for model-based development. Therefore, this article started out analyzing actual model clones in UML domain models, and proposed a terminological framework, a pragmatic definition, and a clone classification schema adapted from work on source code clones.

In this model a clone detection algorithm and model element similarity heuristics based on a detailed examination of actual model structures are developed. Given that many algorithms are proposed, based on intuitions or analytical arguments alone, It is thought this is a particularly sound approach that increases the validity of this approach. This article also provided formal definitions of models, model fragments, and model clones, and implemented our approach in the MQclone tool (pronounce as “m clone”). The detection quality and runtime of this algorithm and heuristics were validated experimentally.

Clone Stability (March 2011)

Code clones are said to threaten the maintainability of a system -- especially when the system evolves and source code is changed. Whether clones truly increase maintenance effort can be analyzed by comparing the stability of cloned code to the stability of non-cloned code. A previous study found that cloned code is even more stable than non-cloned code and, thus, requiring less maintenance effort -- contrary to the frequently voiced assumption. In this paper, it is described that partial replication and extension of this study using a more detailed measurement and considering different parameters for clone detection. In general, the findings of the previous study were also validated. Furthermore possible reasons are explored to gain a better understanding of the unintuitive results.

Model clone detection based on tree comparison (Dec. 2012)

Model driven development has become a key industry practice. With higher levels of abstraction and advent of domain specific languages, models find their presence in every field. Latest software engineering practices lead to large models which are really hard to design and manage. Significant overlaps in large models are really a matter of concern. Anecdotal evidences suggest that clones in models poses similar threats as in code. The paper introduces an approach to detect clones in UML models. The technique is based on finding similarities between two object oriented diagrams. Firstly, UML models are encoded as XMI files. Sub tree comparison is applied after the XMI file is filtered and represented as a tree. Similarity between two diagram elements in a model is determined and reported as a clone.

Towards understanding of classes versus data types in conceptual modeling and UML (Dec. 2012)

The authors describe that the traditional conceptual modeling and UML take different vague, ambiguous, and apparently incompatible approaches to making a distinction between two different entity types - classes and data types. In this paper, an in-depth theoretical study of these ambiguities and discrepancies is given and a new semantic interpretation is proposed for consolidation.
The interpretation is founded on the premise that populations of the two kinds of entity types are defined in two substantially different ways: by intentional (for data types) and extensional (for classes) definitions. The notion of a generative relationship set is introduced to explain the role of specific relationship types that are used to define populations of structured data types by cross-combinations of populations of the related entity types.

Finally, some important semantic consequences are described through the proposed interpretation: value-based vs. object-based semantics, associations vs. attributes, and identity vs. identification. The given interpretation is based on runtime semantics and allows for fully unambiguous discrimination of the related concepts, yet it fits into intuitive understanding and common practical usage of these concepts.

**Software clone detection: A systematic review (Jan 2013)**

This study reports an extensive systematic literature review of software clones in general and software clone detection in particular.

Existing literature about software clones is classified broadly into different categories. The importance of semantic clone detection and model based clone detection led to different classifications. Empirical evaluation of clone detection tools/techniques is presented. Clone management, its benefits and cross cutting nature is reported. Number of studies pertaining to nine different types of clones is reported. Thirteen intermediate representations and match detection techniques are reported. This study call for an increased awareness of the potential benefits of software clone management, and identify the need to develop semantic and model clone detection techniques. Recommendations are given for future research.

**Development process** has shown that the practical application of model-clone detection is not a straight-forward task besides dealing with large-scale models and providing a suitable notion of similarity that also covers model parts with slight modifications or variation, a pragmatic approach To deal with inevitable false positives is necessary.

Since control-theoretic data models are constructed from a small set of basic elements, the identification of similar parts within these models in general leads to a large number of clones. However, only a small fraction of these clones are relevant with respect to maintainability or reuse issues. Therefore, the treatment of large-scale models, the identification of sufficiently similar clones, and the extraction of relevant clones are decisive challenges, for the practical application of model-based clone detection.

**Clone Detection Methodology**

Firstly select UML Reference and Candidate model which extract internal structure in the form of XML. After that, parse the model and store the internal structure of XML and form internal XML Data Structure. Then compare the parse and store value with Heuristics which means selection criteria matching of clone and the result is obtained which means the clone is detected.

A model clone is a set of similar or identical fragments in a model of the system. Understanding and Identifying model clones are important aspects in software evolution. During the Evolution of the Software product, Cloning is often a strategic means for the same. Software clones are important aspects in software evolution. If a system is to be evolved, its clones should be known in order to make consistent changes. Cloning is often a strategic means for evolution.

As shown below, there are two models i.e. reference and candidate model, so one is selected as reference and other as candidate model. By using both models the clone is detected, i.e. model clone detection with the help of heuristics which means selection criteria matching of clone. After that probability is checked. If the probability is greater than expected value then the clone is detected which means that probability for each clone node in reference model otherwise there is no clone detected.
Below is the Reference model for the Clone detector. The Civil Hospital is a simple abstraction of the Real world hospital. In Reference Model i.e. Civil Hospital Model, there is one Package Diagram i.e. Civil Hospital which include five classes i.e. Person, Staff, Patient, Dentist and Doctor. These classes have attributes and operation. Staff, Patient, Dentist and Doctor are generalized with Person Class which means that it include all the attributes and operation of Person Class. Dentists and Doctors are also generalized with Staff Class.

It has been observed that the techniques for detecting clones work very well for Model clones and to detect software clone for the same. Some Heuristics, such as 60% constraint are still required to begin with clone detection but the system produces adequate results for any number of models.

4. CONCLUSION

Clone detection techniques play an important role in software evolution research where attributes of the same code entity are observed over multiple versions. To successfully create any method or technique for model clones detection all the models have to be studied defined in UML including internal and External Structure of UML. This paper reviews some of the techniques available for the Model Clone Prevention and detection.

5. FUTURE SCOPE

In future, a basis for comparisons of UML Domain Models can also be achieved by this researcher. The process will be done using XML parsing of UML domain Models, with Referenced and Candidate Models.

Following are the further topics researcher can work over

Model Clone Detection with Automated Heuristics.
General Clone Detection work for any type of Clone.
Use case and State Chart Clone Detection in UML Models.
Interaction and Activity Clone Detection in UML Domain Models.
Automated Generation of Models from Code to detect Model Clones.

REFERENCES


Analysis and design of a Bidirectional DC–DC Converter for Super Capacitor

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Abstract. This paper presents a new bidirectional DC/DC converter for super capacitor applications. The proposed converter has a parallel structure in super capacitor side and a series structure in the other side. This structure increases efficiency of the converter. For current sharing in the parallel side of the proposed converter, two different methods are used and compared in this paper: Current balancing transformer (CBT) and two separate inductors (TSI). Simulation and experimental results show performance of the proposed converter.

Key words: current sharing, parallel primary, bidirectional converter, super capacitor, fuel cell.

1. Introduction

Currently fuel cell electric vehicles (FCEV) are considered as an attractive option for future cars because of environmental issues and alternative energy requirements. However, since fuel cell stack has a slow response using an auxiliary energy storage device such as battery or super capacitor (SC) is recommended in the fuel cell (FC) applications [1-3] while the battery has a large energy density and SC has a high power density, FC-battery hybrid and FC-SC hybrid system offer different features. However FC-SC Battery hybrid system in Fig-1 have been shown to have superior features [2-3].

Because of the charge dependent voltage of SC, a bidirectional DC/DC converter is needed for bidirectional power exchange between SC and other parts of the system for different voltage levels [2-4]. Isolated full-bridge converter in Fig. 2 is a common DC/DC converter topology [5-6]. For high power applications, parallel isolated full-bridge converters have been proposed [7]. In fuel cell applications, generally low voltage is required to be boosted to higher voltages. Fig. 3 shows the primary parallel isolated boost converter proposed in [8] which is suitable for high voltage gain applications. This converter is composed of full-bridge stages with parallel primary connections (where current is high and voltage is low) and a single rectification stage with series secondary connection (where current is low and voltage is high). Current sharing is ensured by the series connection of transformer secondary windings and small cascaded current balancing transformer (CBT) on the primary side.

In this paper, the unidirectional converter presented in [8] is modified to handle bidirectional power flow in energy storage applications. For this purpose, the diode bridge. For this purpose, the diode bridge rectifier on the secondary side has been replaced with a full bridge inverter. In addition, a detailed analysis has been carried out comparing two different current balancing configurations. Using two separate inductors (TSI), instead of current balancing transformer (CBT) is recommended due to cost and simplicity. It has been shown that the current sharing performance is similar in both cases.
2. **Bi-directional DC/DC Converter**

Fig. 4 shows the modified bidirectional DC/DC converter suitable for SC applications. The converter includes two full-bridge stages in the primary side and one in the secondary. Two inductors with the same value \( L1=L2 \) are used as boosting elements. This configuration (TSI) can eliminate the requirement of current balancing transformer (CBT). The proposed converter has two discharging and charging operating states. These states will be illustrated in the following sections.

3. **Normal Operation Modes**

The proposed converter can be simply modeled as in Fig. 6a and Fig. 6b for CBT and TSI configurations. Respectively. Upper and lower primary full-bridge stages have been modeled by switches \( Sm1 \) and \( Sm2 \) and secondary side bridge has been modeled by \( Sm \).

![Image](image.jpg)

**A Discharge State**

In the discharging of SC, secondary inverter acts as a rectifier. Considering the gate signals in Fig. 5, two main operation modes can be defined for the converter.

1) Mode 1: Both \( SM1 \) and \( SM2 \) are closed and inductors are charging (Fig. 7).
2) Mode 2: Both \( SM1 \) and \( SM2 \) are open and inductors are discharging (Fig. 8)

Fig. 9 shows waveforms for both inductor configurations. For the same ripple current, inductance of each inductors of TSI topology is twice of inductance of inductor of CBT topology. It can be proven for both cases that

\[
V_o = \frac{2n}{D^2} V_i
\]

\[
I_i = \frac{2n}{D^2} I_o
\]

![Image](image2.jpg)

**Fig 7: Model of Proposed converters in Mode 1 of discharging states a) CBT b) TSI**

**Fig 8: Model of Proposed converters in Mode 2 of discharging states a) CBT b) TSI**

![Image](image3.jpg)

**Fig 9: Waveform of proposed converters in normal mode of operation**
B Charge State
Primary side full-bridge stages act as rectifiers during the charging of the SC. Similar to the discharge state, two main operation modes can be considered in the charge state.

1) Mode 1: Both diodes $D_M$ and $D_M$ are conducting and inductors (or inductor) are discharging (Fig. 10).

2) Mode 2: Both $D_M$ and $D_M$ are open and inductors (or inductor) are charging (Fig. 11).

Fig. 12 shows waveforms for both configurations. Neglecting dead time, it can be proved that (1) and (2) are correct in the charge state. However considering dead time (Fig. 5), it can be written

$$V_e = \frac{2n}{D_i} \frac{2D_o}{D_i} V_i$$

$$I_i = \frac{2n}{D_i} \frac{2D_o}{D_i} I_o$$

Fig 10: Model of Proposed converter in Mode1 of charging state a)CBT b) TSI

Fig 11: Model of Proposed converter in Mode1 of charging state a)CBT b) TSI

Fig 12: Waveform of proposed converters in normal mode of charging state.

4. Extra Operation Modes

In the previous section, it is assumed that both primary side switches are switched at same time. However it is possible that this assumption is not satisfied because of differences in component propagation delays and parasitic elements.

Considering a single boosting inductor (without CBT), when one of the simplified switches in Fig. 6 is ON, all the current $i_1$ passes through the corresponding bridge. However in CBT and TSI configurations, due to the high impedance seen between the two full-bridge structures, the current going through each full-bridge can not change instantaneously. Therefore either CBT or TSI should be included in the topology for proper current balancing.

A. Discharge state

Two extra modes can be defined in discharge state:

1) Mode 3 (Fig. 13): $SM_1$ is ON and $SM_2$ is OFF. When both modeling switches $SM_1$ and $SM_2$ are OFF, but $i_2 > i_1$, this mode also can be created as diode $DM_1$ is ON.

2) Mode 4 (Fig. 14): $SM_1$ is OFF and $SM_2$ is ON. When both modeling switches $SM_1$ and $SM_2$ are OFF, but $i_1 > i_2$, this mode also can be created as diode $DM_2$ is ON.

Extra modes 3 and 4 are generally similar to mode 2; however currents $i_1$ and $i_2$ are not equal in these modes. As an example, assuming a delay for turning $SM_1$ ON, we will investigate the operation of the converter in both CBT and TSI configurations.

1) CBT: When both switches $SM_1$ and $SM_2$ are OFF, converter is in mode 2. When $SM_1$ turns ON and $SM_2$ is OFF (due to signaling mismatch), converter has mode 3 (Fig. 13-a). In this mode CBT sees a non-zero voltage; therefore magnetizing current of CBT is increasing. This current creates a current difference between two primary inverters. With turning $SM_2$ ON, converter goes to mode 1, while there is a constant current difference between two primary bridges. When both switches turn OFF again (it is assumed no delay for turning OFF), converter can not enter to mode 2 because inverter currents are not same. Since $i_1$ is greater than $i_2$, simplified diode $DM_2$ is forced to be ON and thus converter will be in mode 4. During mode 4, CBT sees
negative non-zero voltage and magnetizing current of CBT is decreasing. Mode 4 continues until both inverter currents are the same. The required time interval for this is equal to the switching delay time interval between the two switches, $SM_1$ and $SM_2$. After mode 4, converter goes to mode 2. Fig. 15-a shows the waveforms of CBT topology in this condition.

2) TSI: When both switches $SM_1$ and $SM_2$ are OFF, converter is in mode 2. When $SM_1$ turns ON and $SM_2$ are OFF (due to signaling mismatch), converter is in mode 3 (Fig. 13-b). In this mode inductors see different voltages; $L_1$ sees positive voltage and $L_2$ sees negative voltage. This creates a current difference between the two inductors. With $SM_2$ turning ON, converter goes to mode 1, while there is a constant current difference between the two primary bridges. When both turn OFF again, converter can not enter to mode 2 because inverter currents are not same. Since $i_1$ is greater than $i_2$, simplified diode $DM_2$ is forced to be ON and thus converter will be in mode 4. During mode 4, $L_1$ sees negative voltage and $L_2$ sees positive voltage. Therefore the current difference between the inductors is decreasing. Mode 4 continues until both bridge currents are the same. The required time interval for this is equal to the switching delay time interval between the two switches, $SM_1$ and $SM_2$. After mode 4, converter goes to mode 2. Fig. 15-b shows the waveforms of TSI topology in this condition. For both configurations, it can be proven that

$$V_o = \frac{2n}{D_i + d_{on}} V_i$$  \hspace{1cm} (5)
$$I_i = \frac{2n}{D_i + d_{on}} I_o$$ \hspace{1cm} (6)

For both configurations, it can be seen from Fig. 15 that average of currents $i_1$ and $i_2$ are not equaled when there is a switching delay. Defining current difference $I_d$ as

$$I_d = 0.5(I_1 - I_2)$$ \hspace{1cm} (7)

Therefore for CBT and TSI configurations:

it can be written that:

$$I_d_{\text{CBT}} = \frac{D d_{on} V_i}{2 f_{sw} L (D_i + d_{on})}$$ \hspace{1cm} (8)
$$I_d_{\text{TSI}} = \frac{D d_{on} V_i}{2 f_{sw} L (D_i + d_{on})}$$ \hspace{1cm} (9)

Therefore for $L=L_1$, both configurations have similar current difference. However assuming $d_{on}$ is small, current difference ($I_d$) is small in both configurations.

![Fig 13: Model of proposed converter in mode 3 of discharging state a) CBT b) TSI](image)

Fig 13: Model of proposed converter in mode 3 of discharging state a) CBT b) TSI

![Fig 14: Model of proposed converter in mode 4 of discharging state a) CBT b) TSI](image)

Fig 14: Model of proposed converter in mode 4 of discharging state a) CBT b) TSI

![Fig 14: Waveform of proposed converter in discharging state with switching delay](image)

Fig 14: Waveform of proposed converter in discharging state with switching delay

B: Charging State

Similar to discharge state, two extra modes can be defined for charge state

1) Mode 3 (Fig. 16): $DM_1$ (or $SM_1$) is ON and $DM_2$ is OFF.

2) Mode 4 (Fig. 17): $DM_1$ is OFF and $DM_2$ (or $SM_2$) is ON.

Similar to discharge state, extra modes 3 and 4 are generally similar to mode 2; however currents $i_1$ and $i_2$ are not equal in these modes. In charge state, if switching delays are smaller than the dead time, extra modes does not occur. However, extra modes can occur due to the difference between inductor values.
5. Control System

A hybrid system such as in Fig. 1 usually includes a central control system (CCS) and several local control subsystems (Fig. 17-18). Regarding system condition, CCS determines references signals for the control subsystems.

In this paper, it is assumed that the reference current of SC is determined by CCS. Fig. 19 shows the control subsystem for the proposed converter for an SC application. Control system is based on average current mode control. A PI controller has been used for ensuring the steady state current error to be zero. Output of PI is a duty cycle value that it is applied to the PWM block.

![Control system of FC-SC-Battery hybrid system (Fig. 1)](image)

![Control subsystem of proposed bidirectional converter (supercapacitor)](image)

6. Simulation and Experiments

To verify the proposed converter, the control system shown in Fig. 18 was used for simulation. Table 1 shows simulation parameters. A step signal has been used as the reference current; it changes from -100A to 100A at t=2ms. Before t=2ms the converter will be in the charge state and after that converter should go to the discharge state. Simulation results have been shown in Fig. 19. It can be seen that the SC current (I) is able to track the reference current signal in both charge and discharge states.

An experimental prototype of the modified bidirectional DC-DC converter was also built using digital signal processor (DSP) control (Fig. 20). Experimental current I has been shown in Fig. 21. It could be observed that that the closed loop controller implemented in the DSP is able make the SC current follow the reference current in a stable manner. Implementation of the two transformers with primary parallel and secondary series connection is realized with planar E-type cores.

<table>
<thead>
<tr>
<th>Table I. - Simulation Parameters</th>
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</thead>
<tbody>
<tr>
<td>Parameters</td>
</tr>
<tr>
<td>--------------------------------</td>
</tr>
<tr>
<td>Capacitance of SC</td>
</tr>
<tr>
<td>Internal resistance of SC</td>
</tr>
<tr>
<td>Initial voltage of SC</td>
</tr>
<tr>
<td>Battery voltage (V_b)</td>
</tr>
<tr>
<td>Proportional gain (K_p) of PI</td>
</tr>
<tr>
<td>Integral gain (K_i) of PI</td>
</tr>
<tr>
<td>Transformer turns ratio (n)</td>
</tr>
<tr>
<td>Inductors</td>
</tr>
<tr>
<td>Switching Frequency</td>
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</table>

![Simulation results: Current and voltage of super capacitor](image)
7. Conclusion

Here an isolated bidirectional DC/DC converter has been proposed for super capacitor applications. The proposed converter uses a parallel structure on the primary side (low-voltage high-current side) and a series structure in secondary side (high-voltage low-current side). This structure has already been used for unidirectional power flow in the literature. It has been developed for bidirectional power flow in this paper. Also a new method was proposed for current sharing on the primary side. In the proposed method, separate inductors can be used instead of the conventional method which is based on a current balancing transformer. Although both methods have almost the same performance, manufacturing the separate inductors is easier.

References


Designing Of An Isolated Full Bridge Dc-Dc Converter

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Abstract: A new isolated full-bridge DC-DC converter with bi-directional power flow is proposed. By taking some additional auxiliary active clamping circuits to both bridges, zero-voltage and zero-current-switching are achieved to improve the performance of the bi-directional PWM converter. The switches are controlled by phase shifted PWM signals by duty cycle. The principle of operation is analyzed and simulated.

1. INTRODUCTION

In hybrid electric vehicles electric power distribution systems operate at different voltage levels due to the availability of storage devices and Electric motors for traction and fuel cells are connected by a high voltage bus. while batteries and ultra capacitors are connected by a low voltage bus. Bi-directional converters controlling the energy flow between these energy sources are thus required. so high power full-bridge bi-directional converter has become an important research topic during recent years [1] [2].

The current-fed full-bridge and the voltage-fed full-bridge are the essential structure parts. Two voltage-fed full-bridges are the sub circuits of the Dual Active Bridge (DAB), which contain less components as the circuit under investigation and ZVS is achieved in the resonant range [3]. The disadvantage of this circuit is that the performance of The simplest way is by using a RCD snubber to clamp the voltage, but a lower efficiency is the resulting drawback. An “active commutation” control principle was published in [4] in order to control the current of leakage inductance; however clamping circuits are additionally requested. A buck converter or a fly back converter is employed in industrial use in order to replace RCD snubber, but the still needed clamping circuits presented in [8] and [10] are too complex. Considering the boost operational mode, the leakage inductance of the transformer must be very small. The energy stored in the leakage inductance is not sufficient to realize ZVS for the lagging leg. A simple active clamping circuit is employed in [5] and [11], which suits for bi-directional converters [6]. Unluckily, the resonant current increases the current stress of switches largely. the converter is determined by the parameters of the transformer, due to the leakage inductance is used for storing and transferring energy. The freewheeling currents increase the losses and reduce the effective duty cycle.

An alternative topology is shown in Fig. 1, consisting of a current-fed full-bridge at the lower voltage side and a voltage-fed bridge at the high voltage side. Inductor L performs the output filtering, when the energy flows from high voltage bus to batteries or ultra capacitors it described as BUCK mode; basically it works as boost inductor, when energy is provided by the storage element, which is BOOST mode. Advantages of this topology are: The freewheeling current is small, and the leakage inductance is only used as energy transferring element, thereby a largely simplified transformer results when compared with DAB. But the transformer leakage inductance may still cause high voltage transients across the bridge switches due to the current feeding nature, which increases the switching stress and decreases the reliability.

In this contribution an improved full-bridge converter shown in Fig.1 is proposed, where by means of a simple auxiliary circuit at the voltage side, an active clamping circuit consisting of $S_{aux}$ and $C_{cl}$ on the current-fed side $ZVZCS$ is achieved for voltage-fed side switches and, by delaying the turn-off time of
switches S1 and S3, and by adding a simple auxiliary circuit consisting of \( P_{ux} \) and two diodes at the voltage side the switches at current-fed side are operated by ZVZCS. Moreover, the resonant current between the clamping capacitor and leakage inductance is limited, thereby improving the converter efficiency.

2. PRINCIPLE OF CIRCUIT OPERATION

Diagrams are shown in Fig. 2 and Fig. 3. The phase shifted PWM control act as control for both modes. In boost mode, switches S1–S4 are controlled, and the diodes of switches P1–P4 are used as rectifier. in buck mode, switches P1–P4 are controlled, and the diodes of switches S1–S4 operate as rectifier. To simplify the steady-state analysis, several assumptions are made as follows:

1. All components are ideal. The transformer is treated as an ideal transformer and a leakage inductance.
2. Inductor \( L \) is large enough to keep the current \( I_L \) constant during a switch period in both working modes.

a. Operation of Buck converter

The leading-leg switches P1 and P3 are operated at ZVS and the lagging-leg switches P2 and P4 are operated at ZCS by using auxiliary switch and clamping capacitor. This is appropriate for converters, in which the leakage inductance of the transformer is very small \((L_i<1 \, \text{mH})\).

b. Operation of Boost converter

As above explanation by cooperating auxiliary switches \( S_{aux} \) and \( P_{ux} \) shown in Fig. 1, the lower-leg switches S2 and S4 are operated at ZVS, and the upper-leg switches S1 and S3 are operated at ZCS, moreover the voltage stress of the switches are limited.

The operation stages are shown in Fig. 4. Mode 0 \([t < t_0]\): S1, S2 are conducted, so boost inductor L is charged, and the transformer is shorted. S4 and \( P_{ux} \) are turned on, due to the bridge is shorted also by S1 and S2, S4 is turned on at ZVS.

Mode 1 \([t_0 < t < t_1]\): At \( t_0 \), S2 is turned off. The parallel diode of switch \( S_{aux} \) conducts by freewheeling of boost current \( I_L \), \( C_{cl} \) and \( L_{lk} \) resonate. At \( t_1 \), \( i_{Ccl} \) is zero and \( i_{LLk} \) equals \( I_L \).

Mode 2 \([t_1 < t < t_2]\): After \( t_1 \), \( P_{ux} \) is still turned on, \( i_{LLk} \) is freewheeling.

Mode 3 \([t_2 < t < t_3]\): At \( t_2 \), \( P_{ux} \) is turned off, \( S_{aux} \) is turned on. The energy is delivered to the voltage-fed side. In this stage the clamping capacitor \( C_{cl} \) is discharged, the snubbed energy is then delivered to voltage-fed side.

Mode 4 \([t_3 < t < t_4]\): At \( t_3 \), \( S_{aux} \) is off. The energy is still delivered to the voltage-fed side. And in this mode, \( i_{LLk} \sim I_L \).

Mode 5 \([t_4 < t < t_5]\): At \( t_4 \), S3 is turned on by ZCS due to the leakage inductance. After S3 turned on, \( i_{LLk} \) is decreased to zero by the reflected voltage from voltage-fed side.

Mode 6 \([t_5 < t < t_6]\): After \( i_{LLk} \sim 0 \), S1 can be turned off by ZCS. The boost inductor L is
charged. At $t_6$, the next half period begins.

\[
U_{\text{low}} \quad \text{(a) Mode 0 \quad [t < t_o]}
\]

\[
U_{\text{low}} \quad \text{(b) Mode 1 \quad [t_0 < t < t_1]}
\]

\[
U_{\text{low}} \quad \text{(c) Mode 2 \quad [t_1 < t < t_2]}
\]

\[
U_{\text{low}} \quad \text{(d) Mode 3 \quad [t_2 < t < t_3]}
\]

\[
U_{\text{low}} \quad \text{(e) Mode 4 \quad [t_3 < t < t_4]}
\]

\[
U_{\text{low}} \quad \text{(f) Mode 5 \quad [t_4 < t < t_5]}
\]

\[
U_{\text{low}} \quad \text{(g) Mode 6 \quad [t_5 < t < t_6]}
\]

Fig.4 Operation stages of boost mode.

**DESigning PROCESS**

a. Soft switching condition

1. Current-fed side

The lower-leg switches S2 and S4 are turned on at ZVS in any load condition, due to the current-fed bridge is shorted before S2 and S4 are turn on. The delay between the upper-leg current-fed switches must be longer than interval $[t_5 < t < t_6]$.

The minimum delay time is:

\[
T_{d \min, S1S2} = \frac{L_{\text{in}} I_L}{V_{\text{CD}}} \quad (1)
\]

2. Voltage-fed side

The dead time for leading-leg switchs are determined by the load and the input voltage, the minimum dead time must be as long as the capacitor resonance time:

\[
T_{d \min, P1P2} = (C_{p1} + C_{p2}) \frac{U_{\text{high}}}{nI_L} \quad (2)
\]

For the lagging-leg switches ZCS operation is determined at the maximum duty cycle. Before turned-on the lagging-leg switches, the leakage inductance current must be zero. So the reserved time is

\[
T_{r \min, P2P4} = \frac{L_{\text{in}} I_L}{V_{\text{CL}}} \quad (3)
\]

3. Clamping capacitor

Due to discharging of the leakage inductance the energy stored in the clamping capacitor must be satisfied with the maximum required energy, so

\[
C_{cl} \geq \frac{L_{\text{in}} I_L^2}{V_{\text{CL}}^2} \quad (4)
\]

B. Reduction of the snubbed energy

In the interval $[t_0 < t < t_1]$, the high transient voltage occurs inevitably in boost mode, which could be snubbed by the clamping branch ( $S_{aux}$, $C_{cl}$ ). The snubbed energy of $C_{cl}$ is transferred to the voltage-fed side via switches S1~S4, hence a minor increase of current result for switches. The auxiliary switch $S_{aux}$ is also employed to reduce the snubbed energy; subsequently, the current stress can also be reduced.
The current of inductor L is employed as current feedback. references are given by the DSP to the current compensators.

\[
\begin{align*}
I_{CL} &= I_L \cos(\theta t) + \sqrt{C_L / L_L} (U_{C_L0} \sin(\theta t)) \\
I_S &= I_L + I_{CL} \\
\text{With} & \quad \theta = 1 / \sqrt{C_L L_L}
\end{align*}
\] (5)

When \( P_{in} \) is conducting, the transformer is clamped. Hence the voltage \( V_T \) equals zero, the leakage inductance current \( I_S \) increases sharply, and the clamping capacitor current decreases quickly. The energy flowing through clamping circuit is reduced and can be expressed as:

\[
W_{CL} = 2U_{C_L0} (I_L \sin(\theta t)) + \sqrt{C_L / L_L} U_{C_L0} \cos(\theta t)
\] (6)

As the commutation interval is reduced, the dynamic response and the efficiency of the filter is increased.

(4) Controlling process

The controlling process is shown in Fig. 6, which is integrated with the control for boost and buck mode operation together, thus the cost for controlling section is reduced.

The voltage control and average current control are used here to control the energy flow and to improve the converter dynamic behavior and stability. The current control loop of boost mode and buck mode are implemented by using analog current compensator and PWM modulator, while the voltage control loop is realized using a DSP controller.

The DC-bus voltage and the voltage of the energy storage element are sampled by the DSP controller as voltage feedback. According to the working mode, the situation of the load and energy storage element, the current references are given by the DSP to the current compensators. The current of inductor L is employed as current feedback. There are two phase shift PWM IC UCC3895 used as PWM modulator to generate the gate driver signals.

The gate drive signals of the auxiliary switches shown in Fig. 2 and Fig. 3 are generated by a CPLD, which is employed also to monitor the fault signals of the converter and to perform the protection schemes. response and the efficiency of converter will be increased.

SIMULATION ANALYSIS

To verify the theoretical analysis of the proposed topology, simulation models are built in SIMPLORER using the following design specification: \( U_{\text{Low}} \geq 40 \text{V} \), \( U_{\text{High}} \geq 650 \text{V} \), \( L_B \geq 0.5 \text{mH} \), \( L \geq 23 \text{mH} \), \( C_{CL} \geq 2 \text{uF} \), and the switching frequency \( f_S \geq 50 \text{kHz} \).

Simulation results of buck mode are shown in Fig. 7, and Fig. 8 displays the boost mode simulation results. The transformer primary side voltage \( u_{AB} \) and secondary side voltage \( u_{CD} \), leakage inductance current \( I_{LB} \), clamping capacitor current \( I_{CL} \) and voltage \( U_{CL} \) are displayed.
From Fig. 8 we see that the simulation results agree well with the theoretical waveforms of Fig.3. With the functions of the auxiliary switch P aux the resonant current between clamping capacitor CCl and leakage inductance Lrk is reduced, thus the current stress of switches are also reduced.

5. EXPERIMENTAL RESULTS

A 3kW experimental prototype is built to verify the operation principle of the proposed converter. A super capacitor module working at the low voltage side is employed as energy storage element, whose voltage range is 20~42V. The high voltage range is 500~750V. The parameters of circuit are listed as follow:

<table>
<thead>
<tr>
<th>Component</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>P ~ P</td>
<td>IGBT</td>
</tr>
<tr>
<td>Dp1 ~ Dp4</td>
<td>IGBT</td>
</tr>
<tr>
<td>aux</td>
<td>MOSFET</td>
</tr>
<tr>
<td>S1 ~ S4</td>
<td>MOSFET</td>
</tr>
<tr>
<td>Saux</td>
<td>snubber capacitor</td>
</tr>
<tr>
<td>Ccl</td>
<td>2.4* F</td>
</tr>
<tr>
<td>n</td>
<td>transformer turns ratio</td>
</tr>
<tr>
<td>Lrk</td>
<td>0.5* H</td>
</tr>
</tbody>
</table>

The experimental waveforms of the input inductor current, the current-fed side transformer current and the voltage across the transformer at boost mode are shown in Fig. 9. The test waveforms of the clamping circuit are shown in Fig. 10. The output power is 550W in this test and the measured efficiency is 90.5%.

From the experiments, it can be seen that the voltage peaks caused by transformer leakage inductance are reduced by the clamping circuit.

![Fig. 9 Boost mode tests.](image)

Ch1: Inductor current iL (10A/div).
Ch2: Current-fed side transformer current iLrk (10A/div).
Ch3: VAB.
Ch4: VCD.

![Fig. 10 Clamping circuit test at boost mode. Ch1: Voltage of clamping switch Saux. Ch2: Current of clamping capacitor Iccl (5A/div). Ch3: VAB. Ch4: VCD.](image)

At the same time, the current stress of the current-fed switches are decreased. The resonances of VAB are mainly caused by the leakage inductance and the parallel capacitors of the voltage-fed switches.
CONCLUSION
In this paper, a new isolated full-bridge bi-directional
DC-DC converter using phase shifted PWM control is
presented. By using clamp circuits at current-fed and
voltage-fed side, the voltage transient voltage across the
circuit-fed bridge are limited, and all switches are operated
by soft switching The operating principle and designing
process is discussed and verified by simulations and
experiment.

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A Novel isolated Bi-directional DC-DC converter for Renewable energy storage systems

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Abstract- To improve the energy quality and store the energy most of the renewable energy systems include an energy storage element are charged by the bidirectional DC-DC converter. This paper presents a bidirectional DC-DC converter which employs the two bridge, resonant class-E converters on the both sides of the isolating transformer. The small side converter is controlled as step up and the high side converter is controlled as step down. The proposed strategy is characterized by good dynamic properties, because the converter transistors are switched in zero voltage switching conditions. A theoretical analysis to provide relations for system design, and the laboratory model investigations to validate the system characteristic is presented.

Key words: Isolated bidirectional dc − dc converter, RESS (Renewable energy storage systems).

1. Introduction

The essential part of the renewable energy system is a storage element [1–6]. The storage element gathers the energy fluctuations and enables to improve the system dynamic properties. A chemical battery or a super capacitor, used as a typical energy storage element, are characterized by the small nominal DC voltage value. To charge and discharge the storage element, the bidirectional DC-DC converter is used. The DC-DC converter provides an electrical isolation between small voltage and high voltage parts of the system, and then the transformer is used for isolation. In order to feed the transformer a DC power must be converted into AC power and rectified to DC power. To minimize the transformer size, weight and cost, the frequency of the AC power should be high. The frequency increase is limited by the transistor conduction and switching losses. It should be noticed that the main source of the power dissipation is the small voltage side converter because it conducts a high current. So, the main goal of the research is directed to the small voltage converter efficiency. The first proposal of the bidirectional DC-DC converter system was a DAB (dual active bridge) converter [7]. The DAB converter consists of the two voltage-fed inverters at each side of the transformer. The energy flow value and direction were controlled by the phase-shifting angle of the both inverters. The main drawback of the DAB converter is that it does not accept a high difference between voltages of small and high sides of transformer, because then the current stress and losses caused by the circulating current become to high. Additionally this system does not ensure ZVS conditions of the transistors switching process in a wide range of the voltages variations.[8] consists of a current-fed (boost) inverter at a small voltage side and a voltage-fed (buck) inverter at a high voltage side. The drawback of this system is the transient provoked by the transformer leakage inductance when the boost converter is switched. transformer leakage inductance can be used as a useful element in the resonant converters [9]. In the paper [10] a bridge configuration class-E buck resonant converter is proposed. The bridge configuration class-E boost resonant converter was proposed in [11]. The class-E converters guarantee ZVS switching conditions for converter transistors in the whole operating range and apart from that do not generate oscillations which invoke the voltage transient. On the basis of both class-E converters mentioned above a bidirectional class-E DC/DC converter is covered.

2. Topology, control principle and modes of operations

The proposed converter topology is given in Fig. 1. To the energy flow from the small to the high voltage side, the boost converter (L) is controlled and the high side converter (H) is not controlled but operates as a rectifier. To the energy flow into the opposite side the buck converter (H) is controlled and the small side converter (L) operates as a rectifier. The main problem of this solution is the use of the same resonant circuit elements for the both directions of the energy flow. As the authors’ theoretical investigation shows, such a situation is impossible and an additional capacitor must be used when the system operates in the buck mode. This capacitor C is joined by the additional switch given in Fig. 1.

The converter system is controlled by the transistors control pulse frequency change and maintaining a constant brake between pulses, the
transistor control pulse width is controlled. The minimum value of the control frequency corresponds to the maximal transistor control pulse width and the maximal value of the energy flow. As we can see in Fig. 2 the transistor control pulse width of the buck converter can be changed from the minimum value.

![Fig. 1. Circuit of Proposed bidirectional DC-DC converter topology](image1)

where \( I_m \) is the amplitude of the high side transformer current

![Fig. 2. The control pulses of buck converter (H)](image2)

The boost transistors control pulses width (frequency) which is controlled within the range when control pulses of a cycle (Fig. 3).

![Fig. 3. The control pulses of boost converter (L) transistors](image3)

The both converters are designed for an optimal operation point. The optimal operation point corresponds to the minimum pulse width (maximal frequency) of the buck converter and to the maximal pulse width (minimum frequency) of the boost converter. During the optimal operation the converter current is flows from the transistors. Transistors current waveforms have only positive value.

### 3. Steady state analysis and design guidelines

#### 3.1. The buck’s working principle. The characteristic wave-forms of the buck mode 2πf of the optimal operation point are given in Fig. 4.

Assuming that the converter (H) input current \( I_h \) is con-stant and the converter output \( i_h \) has sinusoidal shape and due to the converter configuration symmetry, the converter leg current can be expressed as:

\[
i_h = \frac{m}{2} \sin \omega t + \frac{h}{2},
\]

(1)

The part of this current charges and discharges the capacitor \( C_h \) from zero voltage to zero voltage, so in accordance with Fig. 4

\[
\frac{m}{2} \sin \omega t + \frac{h}{2} d(\omega t) = 0,
\]

(2)

From the above equation we can find the relation between the converter input and output currents as a function of the transistor control pulse width.

\[
I_m = \frac{m \sin \lambda + \pi}{1 - \cos \lambda}.
\]

(3)

Then the instantaneous value of capacitor \( C_h \) voltage can be calculated.

In the steady state operation, the mean value of capacitor \( C_h \) voltage is equal to the half value of the high voltage \( V_h \)

\[
\frac{h}{2} = \frac{2 \pi - \phi}{2} V_h d(\omega t).
\]

(4)

So we can find the equation to calculate the desired value of the capacitance \( C_h \)

\[
C_h = \frac{2 \pi - \phi}{2 \pi} \left[ \cos (\lambda - \phi) - (\lambda - \phi) \sin \phi \right] + \left[ \cos (\lambda - \phi) \sin \phi \right] - \sin \phi (2 \pi - \lambda) + \sin \lambda.
\]

(5)

To find the desired capacitance \( C \) value we introduce the folsmalling procedure. If the capacitor \( C \) voltage waveform is sinusoidal and has amplitude \( V_m \) then the rectifier (L) input current \( I_{L} \) rms value can be calculated as:

\[
I_{LR} = \frac{2P_{min}}{V}.
\]

(6)

In the steady state operation the rectifier output voltage value is equal to the battery voltage \( V_b \) and, because of sinusoidal shape of the rectifier input voltage, the folsmalling equation is fulfilled:

\[
V_m = \pi V_b.
\]

(7)

Using Eqs. (6)-(7) we can find the desired capacitance \( C' \)
3.2. The boost operation principle. The direction of the energy flow from the small voltage to the high voltage sources is called the boost operation mode. In this mode the converter (L) is controlled and converter (H) is operated as rectifier. The system design concerns the optimal operation point when the maximal energy value is transferred and the transistors control pulses have a maximal width and overlap, and now only the guideline for its design is giving.

When we assume that the optimal operation point corresponds to $I_{\text{m}} = 0.26$ then $I_{\text{as, lm}}$ and capacitance $C_{1}$ can be calculated as: $I_{\text{m}}$

$$C_{1} = 8.866 \cdot 10^{-3} \frac{3}{C_{\text{1}} L}$$

The last desired element, the resonant circuit inductance $L$ value can be calculated using Folsmalling equation:

$$1.1T = 2\pi \sqrt{\frac{L}{C_{\text{1}} L}}$$

Described Eqs. (6), (7), (8), (9) are sufficient basis for the converter design.

4. Experimental Solution

The proposed bidirectional DC-DC converter prototype was built and tested in experimental circuit given in Fig. 5.

The $V_{1} = 51 \text{ V}$ was used as small voltage sources. As the high voltage $V_{H} = 330 \text{ V}$ source was used DC motor coupled with AC motor transferred energy to the grid $3 \times 230 \text{ V}$. The converter circuit parameters are summarized in Table 1

![Characteristic waveforms of the boost optimal operation point, 1) transistor control pause, 2) transistor voltage, 3) transistor current, 4) the transformer high side current](image)

Waveforms show the characteristic point of operations. The optimal operation point when the maximal energy value is transferred from the accumulator to the grid is presented in Fig. 6.

![Characteristic waveforms of the buck optimal operation point, 1) transistor control pause, 2) transistor voltage, 3) transistor current, 4) the transformer high side current](image)

![Characteristic waveforms of the buck optimal operation point, 1) transistor control pause, 2) transistor voltage, 3) transistor current, 4) the transformer high side current](image)

The next optimal operation point when the minimum power value is transferred from the grid to the accumulator is presented in Fig. 7.

For the both situations the transistor current waveforms have only positive value. The ZVS transistor switching process is evident because the transistors’ current and voltage waveforms do not overlap. The transformer current has sufficiently sinusoidal waveforms, as we have assume. Oscillation visible on the transistor current and voltage. The current probe measured the
transistor, it is located on the connection of the transistor with the external capacitor and it introduces inductance. But, the bidirectional converter system input and output current are without any oscillations, that is important for the energy storage element.

When the transferred energy in boost directions has smaller than maximal value or the transferred energy in buck direction is higher than the minimum value, the converter operates in sub-optimal operation mode.

The waveforms are presented in Fig. 8 for the boost operation and in Fig. 9 for the buck operation.

The transistor current waveform has positive and negative value. Because the transistor current and voltage wave-forms do not overlap, the ZVS conditions are fulfilled. Bi-directional DC-DC converter is devoid of the transistors switching power dissipation within the whole operation range.

Efficiency curves measured on the experimental setup are given in Fig. 10 and the transferred power in Fig. 11.

Due to the high converter switching frequency (200 kHz ÷ 300 kHz), the converter size and weight are minimized and dynamic properties are attractive.

![Fig. 8. Characteristic waveforms of the boost sub-optimal operation point, 1) transistor control pulse, 2) transistor voltage, 3) transistor current, 4) the transformer high side current](image8)

![Fig. 9. Characteristic waveforms of the buck sub-optimal operation point, 1) transistor control pulse, 2) transistor voltage, 3) transistor](image9)

![Fig. 10. The power transferred efficiency, L-H – boost operation, H-L – buck operation](image10)

![Fig. 11. The transferred power, H-L – buck operation, L-H – boost operation](image11)

### 5. Conclusions

This paper presents the novel class E buck/boost resonant bidirectional DC-DC converter for renewable energy system. Important features of the presented converter topology are small size, less weight and high dynamics because of the transistors ZVS switching process with high frequency. The converters employed in the system are current sourced. System is environmentally friendly.
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A Novel Design Of A Bi-Directional DC-DC Converter For A Regenerative Energy Storage System

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Abstract—This paper presents a bi-directional dc-dc converter suitable for an energy storage system with an additional function of galvanic isolation. An energy storage device such as an electric double layer capacitor is directly connected to one of the dc buses of the dc-dc converter without any chopper circuit, the dc-dc converter can continue operating when the voltage across the energy storage device droops along with its discharge. Theoretical calculation and experimental measurement reveal that power loss and peak current impose limitations on a permissible dc-voltage range. This information may be useful in design of the dc-dc converter. A laboratory model of the energy storage system rated at 220 V and 2.6 kJ designed and constructed in this paper verifies that the dc-dc converter can charge and discharge the capacitor bank properly, the dc-dc converter can charge the capacitor bank from zero to the rated voltage without any external circuit.

1. INTRODUCTION

Generally, electric power generated by renewable energy sources is unstable in nature, thus producing a bad effect on the Fig. 2. A bi-directional isolated dc-dc converter, utility grid. This fact spurs research on energy storage systems to smooth out active-power flow on the utility grid [1], [2]. Fig. 1 shows a conventional energy storage system employing a line-frequency (50- or 60-Hz) transformer, a PWM converter, a bi-directional chopper, and an energy storage device such as electric double layer capacitors (EDLCs). The transformer is used for some applications that require voltage matching and for galvanic isolation between the utility grid and the energy storage device. Replacing the line-frequency transformer with a high-frequency and isolated dc-dc converter will result in a more compact and good energy storage system

Various bi-directional isolated dc-dc converters have been proposed as the interface to energy storage devices with focus on automotive or fuel cell applications. Most of the presented dc-dc converters have asymmetrical circuit, several tens volts and several hundreds volts [3]-[9].

Fig. 2 shows a bi-directional isolated dc-dc converter presented in 1991 [10], [11]. It had two symmetrical single-phase voltage-fed full-bridge converters. The dc-dc converter suffered from a low efficiency because the first-generation IGBTs were used as switching power devices [10]. However, advancement in power device technology over the last decade has enabled the dc-dc converter to operate at an efficiency as high as 97% [12]. A resonant dc-dc converter based on the similar topology has also achieved the same efficiency [13]. In addition, when the SiC power devices become available in the near future, the efficiency of the dc-dc converter in Fig. 2 may reach 99%. Thus, the dc-dc converter in Fig. 2 has become a promising candidate as a power electronic interface for an energy storage system.

Fig. 3 shows the energy storage system using the bi-directional isolated dc-dc converter in Fig. 2. By choosing the transformer turn ratio \( n \) enables to design the voltage range of the energy storage device, independent of the utility voltage. The energy storage device is directly connected to one of the dc buses of the dc-dc converter without any chopper circuit. The dc-dc converter continues operating even when the voltage across the energy storage device, \( V_{D2} \) droops along with its discharge.

There, no paper has addressed the permissible voltage range of \( V_{D2} \) in terms of power loss and peak current. There has been no experimental verification based on the dc-dc converter. This paper analyzes the relationships between the...
power loss, the peak current, and $V_{D2}$ in a dc-dc converter rated at 11 kW and 21 kHz with $V_{D1}$ fixed to 325 V. Then, the dc-dc converter is constructed and experimentally tested to verify the analysis. A 2.6-kJ laboratory model of the energy storage system using an electrolytic capacitor bank, together with the dc-dc converter, demonstrates stable charging and discharging operation. Besides, the dc-dc converter can charge the capacitor bank from zero to the rated voltage without any external precharging or starting-up circuit.

2. THE BI-DIRECTIONAL ISOLATED DC-DC CONVERTER

A. Operation Principle and Simplified Theoretical Waveforms

Fig. 4 shows theoretical waveforms of the dc-dc converter where $V_{D1} < V_{D2}$. The two single-phase voltage-fed full-bridge converters produce square voltages $v_1$ and $v_2$. The power transfer $P_D$ can simply be controlled by adjusting the phase shift between $v_1$ and $v_2$, $\delta$ as expressed by [10]

$$P_D = \frac{V_{D1} V_{D2}}{\omega L} \frac{\delta - \frac{\pi}{2}}{\pi},$$

where $\omega (= 2\pi f)$ is the switching angular frequency of the two single-phase voltage-fed full-bridge converters, and $L$ is the sum of the transformer leakage inductance $L_{\text{trans}}$ and the inductance of the auxiliary inductors $L_a$.

Fig. 4, this paper shows a set of two instantaneous values of the current $i_1$ as “switching currents,” $I_{11}$ and $I_{12}$ which are calculated as

$$I_{11} = -\frac{(V_{D1} + V_{D2})\delta + (V_{D1} - V_{D2})(\pi - \delta)}{2\omega L}$$

and

$$I_{12} = \frac{(V_{D1} + V_{D2})\delta - (V_{D1} - V_{D2})(\pi - \delta)}{2\omega L}.$$ 

$I_{11}$ and $I_{12}$ are the instantaneous values of $i_1$ when $v_1$ and $v_2$ respectively change their polarity from negative to positive.

In this paper, a single-phase voltage-fed full-bridge con-verter is referred to simply as a “bridge.” In the following experiments, the transformer turn ratio is unity ($n = 1$) for the sake of simplicity.

B. An Experimental Circuit of the DC-DC Converter

Table I summarizes the circuit parameters of the dc-dc con-verter. Four auxiliary inductors, totally having $L_a = 41 \mu H$, are connected in series with the transformer to obtain an inductance of $L = 42.6 \mu H$ together with the leakage inductance of the transformer, $L_{\text{trans}}$. The inductance of 42.6 $\mu H$ is sufficient to maintain a control resolution of power transfer around 120 W because the time resolution of the controller is 50 ns that corresponds to 0.36$^\circ$ at 20 kHz.

The following sections analyze relationships between power transfer and power losses in the dc-dc converter. The power losses depend not only on the power transfer, but also the dc voltage $V_{D2}$. When $V_{D2}$ droops along with discharge of the energy storage device, power loss increases at a given power transfer.
3. SNUBBER LOSS

A. Operating Points and ZVS Conditions

In Fig. 2, a snubber capacitor $C$ is connected in parallel with each IGBT both to reduce switching loss and to damp out overvoltage. If the IGBT is turned on with its snubber capacitor charged, the capacitor is shorted out by the IGBT, and the energy stored in the capacitor is dissipated, thus resulting in power loss. This paper refers to this power loss as “snubber loss.”

When both dc voltages are equal ($V_D1 = V_D2$), and the power transfer is sufficiently large around its rating, each IGBT is turned on in ZVS (zero-voltage switching) manner to generate no snubber loss. However, when $V_D1 = V_D2$, and the power transfer is small, the IGBT is not necessarily turned on in ZVS manner. Fig. 5 shows simplified theoretical waveforms when the IGBTs in bridge 1 is turned on in hard-switching manner.

The power transfer is less than that in Fig. 4 although the dc voltages $V_D1$ and $V_D2$ are the same as those in Fig. 4. The switching current $I_{11}$ is now positive in Fig. 5 in contrast to negative $I_{11}$ in Fig. 4. With a positive $I_{11}$, the so-called “re-verse recovery” occurs in the freewheeling diodes in bridge 1, leading to hard-switching operation. Turn-on operations of (c), (d) the IGBTs in bridge 1 and bridge 2 can be classified into the following three: (1) ZVS operation, (2) incomplete ZVS operation, and (3) hard switching operation, depending on the phase shift $\delta$, the dc voltages $V_D1$ and $V_D2$, and the dead time $T_d$. The incomplete ZVS and hard-switching operations can take place only in one bridge whose dc voltage is lower than the other. Thus, the four IGBTs in bridge 2 are turned on in ZVS manner because $V_D1 < V_D2$. The following calculations mainly focus on phenomena in bridge 1 because those in bridge 2 can be described alike.

B. Calculations of the Snubber Loss

1) ZVS operation: Fig. 6 shows circuit modes when a leg in bridge 1 (for example, consisting of S1 and S2) operates in ZVS manner. Before the dead time, a current of $I_{11}$ is flowing in S2 (see Fig. 6(a)). When S2 is turned off, the dead time starts. The current flowing in S2 commutates to the snubber capacitors C1 and C2. Resonance between the inductance L (see Fig. 2), C1, and C2 begins. C1 discharges from $V_D1$ to zero while C2 charges from zero to $V_D1$. The energy stored in C1 is transferred to C2. When C1 discharges down to zero, the current commutates to D1 (see Fig. 6(c)). An amount of energy stored in L is regenerated back to $V_D1$ through D1. Providing a gating signal while D1 is conducting can turn S1 on in ZVS manner. This operation results in no snubber loss.

2) Incomplete ZVS Operation: IGBTs in bridge 1 can not necessarily be turned on in ZVS manner even with a negative $I_{11}$. If the magnitude of $I_{11}$, or $|I_{11}|$ is smaller than $I_{\text{min}}$, C1 is not discharged down to zero, and C2 is not charged up to $V_D1$ where [1]

$$I_{\text{min}} = \frac{2V_D1V_D2}{Z_r}$$

Turning S1 on with C1 charged results in an amount of snubber loss. This paper refers to this as “incomplete ZVS operation.” Snubber loss produced by incomplete ZVS operation can be calculated as follows. The collector-emitter voltage of S1, $v_{\text{CE1}}$ in Fig. 6(b) can be expressed as: 
where \( t \) is the time after the beginning of the dead time, and \( \omega r = (1/\sqrt{LC}) \) is the resonant angular frequency of \( C \) and \( L \). At the end of the dead time \( (t = T_d) \), \( v_{CE1}(T_d) \) is not zero because \( |I_{11}| < I_{\text{min}} \). \( C_1 \) is shorted out and quickly discharges from \( v_{CE1}(T_d) \) to zero. \( C_2 \) suddenly charges from \( V_D1 - v_{CE1}(T_d) \) to \( V_D1 \). As a result, a joule loss of

\[
W = C \{ v_{CE1}(T_d) \}^2
\]

is dissipated in \( S_1 \), where \( C = C_1 = C_2 \). Note that charging \( C_2 \) as well as discharging \( C_1 \) contributes to the joule loss. \( W_{\text{snub}} \) represents an amount of energy lost at one switching per leg. The snubber loss \( P_{\text{snub}} \) in bridge 1, having two legs, is calculated as

\[
P_{\text{snub}} = 4 \cdot f \cdot W_{\text{snub}} = 4 \cdot f \cdot C \{ v_{CE1}(T_d) \}^2.
\]

3) Hard-Switching Operation: Fig. 7 shows circuit modes when the leg operates in hard-switching manner. If \( V_{D1} < V_{D2} \), and the following equation is satisfied, the switching

at \( V_{D1} \) (see Fig. 7(a)). Just after \( S_1 \) is turned on, reverse recovery occurs in \( D_2 \). \( C_1 \) rapidly discharges from \( V_{D1} \) to zero, and \( C_2 \) charges from zero to \( V_{D1} \) (see Fig. 7(b)). The charging/discharging currents result in a joule loss of \( W_{\text{snub}} = C \{ V_{D1} \} \) in \( S_1 \). Then, the snubber loss \( P_{\text{snub}} \) in bridge 1 is calculated as

\[
P_{\text{snub}} = 4 \cdot f \cdot W_{\text{snub}} = 4 \cdot f \cdot C \{ V_{D1} \}^2.
\]

As can be seen in (8) and (10), the snubber loss \( P_{\text{snub}} \) is proportional to the capacitance of the snubber capacitors \( C \). Minimizing optimum value inductances in the dc-dc converter circuit is necessary so that small snubber capacitors can damp out the overvoltage appearing across the IGBTs without causing an excessive snubber loss.

IV. PROFILE OF THE CURRENT \( i_1 \) AND RELATED LOSSES

A. Conducting Loss in the IGBTs

This paper approximates both the on-state voltage across the IGBT, \( V_{CE(\text{sat})} \), and the forward voltage drop across the free-wheeling diode, \( V_F \), to be 1.5 V, independently of the current flowing in them [12]. The conducting loss in the IGBTs and diodes, \( P_{\text{cond}} \) can be calculated from the average of the absolute value of the current \( I_1 \), or \( |i_1| \).

When both bridge 1 and bridge 2 is operated in ZVS or incomplete ZVS manner, calculation on Fig. 4 yields

\[
|j_1| = \frac{V_{D1}V_{D2}}{\omega L(V_{D1}+V_{D2})} \left( \frac{\delta^2}{\pi} + \delta + \frac{(V_{D1}-V_{D2})^2}{4V_{D1}V_{D2}} \right) \cdot 1.1.
\]

On the other hand, when either bridge 1 or bridge 2 is operated in hard-switching manner, calculation on Fig. 5 derives:

\[
|j_1| = \frac{1}{\omega L} \left[ \frac{V_{D1}V_{D2} - \delta^2}{V_{D1} - V_{D2}} \right] + \frac{V_{D1} - V_{D2} \pi}{4}.
\]

To calculate \( |i_1| \). \( I_{11} \) and \( I_{12} \) should be obtained first, and then either (11) or (12) should be applied, depending on the switching manner.

B. Copper Loss in the Transformer and the Inductors

The rms value of the switching manner. The copper loss in the transformer and the auxiliary inductors, \( P_{\text{copper}} \) is obtained as current \( I_{11} \) becomes positive, and the IGBTs in bridge 1 are turned on in hard-switching manner: [10]

\[
\delta \leq \frac{V_{D2} - V_{D1}}{2V_{D2}} \pi.
\]

Before the end of the dead time, \( C_1 \) is charged

\[
\frac{V_{CE1}(t)}{2} = \left( V_{D1} + V_{D2} \right) + (V_{D1} - V_{D2}) \cos \omega_r t.
\]

B: Copper Loss in the Transformer and the Inductors

The rms value of the switching manner. The copper loss in the transformer and the auxiliary inductors, \( P_{\text{copper}} \) is obtained as current \( I_{11} \) becomes positive, and the IGBTs in bridge 1 are turned on in hard-switching manner: [10]
The four auxiliary inductors were constructed using ferrite cores. The effective cross-sectional area of each core was $A_e = 3.3$ cm$^2$, the effective volume was $V_e = 37.2$ cm$^3$, and the turn number was $N = 6$. An air gap of $g = 1.5$ mm was introduced in the magnetic path. Thus, the instantaneous magnetic flux density $b_{ind}$ is approximately expressed as $\mu_0$ is the permeability of vacuum. datasheet of PC44 indicates that its core loss per volume is 0.6 W/cm$^3$ when the maximum flux density is 0.2 T at a frequency of 100 kHz in a temperature of 25°C. If the core loss per volume by $k/\mu B^2$, where $k$ is the frequency of magnetization, the coefficient $k = 0.15$ mW/HzT$^2$. This paper assumes that a 20-kHz sinusoidal current having an rms value as large as $I_1$ is responsible for the core loss in the auxiliary inductors. Under this assumption, the core loss in the four auxiliary inductors can be calculated. Thus, the core loss in the auxiliary inductors can be treated as an equivalent winding resistance of

$$R_{core(ind)} = \frac{8k}{\mu_0} \frac{N^2 V_e}{2g} \Omega \quad (14)$$

The core loss in the auxiliary inductors can be calculated as a part of copper loss.

5. POWER LOSSES

A. Comparison between Theoretical and Experimental Losses

Theoretical losses described above are compared to measurement results on the basis of an experimental dc-dc converter rated at 11 kW and 21 kHz. The circuit configuration and the circuit parameters are the same as those in Fig. 2 and Table I, respectively. A regulated dc power supply is connected to the dc bus of bridge 1. The dc bus of bridge 2 is connected back to that of bridge 1 so that the transferred power can be coming from the dc power supply equals the overall loss in the dc-dc converter. Both theoretical calculation and measurement are carried out under $V_{D1} = V_{D2} = 351$ V.

Fig. 8 shows comparisons between the theoretical and experimental losses. The solid line corresponds to the theoretical overall loss, $P_{theory}$ although it excludes the switching loss in the IGBTs, or $P_{SW}$. When $P_D = 11$ kW, the theoretical losses were as follows. The conducting loss was $P_{cond} = 190$ W. The snubber loss was $P_S = 0.0$ W. The copper loss both in the transformer and the inductors $P_{copp} = 75$ W including the core loss in the inductors, $P_{core(ind)}$. The core loss in the transformer was $P_{core(tr)} = 19$ W. The theoretical overall loss $P_{theory}$ was 281 W. The experimental value of the overall loss, on the other hand, was 410 W. Thus, the difference between the theoretical and measurement results was 118 W. It will be correspond to the switching loss in the IGBTs that was excluded from the theoretical overall loss. In [12], the switching loss in the IGBTs was 90 the theoretical calculations above can be valid because the error of 28 W corresponds to 0.28% of the power transfer of 11 kW, and 8% of the measured overall loss of 410 W.
Fig. 9 shows theoretical calculation results of conducting and snubber losses in the IGBTs (Pcond + P snub) when the power transfer pd is positive. In fig 9, one dc voltage VD1 was kept constant at 321V, while the other dc voltage VD2 was changed as a parameter.

By using ZVS operation becomes difficult with VD1 F=VD2, compared to VD=VD2=321V, resulting in an increased snubber loss around 3kW.

Fig. 9 shows Pcond + P snub = 212 W at VD1 = VD2 = 320 V as a “temperature limit.” In the dc-dc converter, the losses in the IGBTs, which are the most dominant in the overall loss, may make the IGBT modules mounted on a heatsink suffer from the highest temperature. The temperature of the IGBT modules, more precisely semiconductor dies in the modules, determines the maximum power transfer. Thus, this paper considers only the losses in the IGBTs as the temperature limit.

When VD2 = 181 V, the losses in the IGBTs exceed the temperature limit at PD > 5.5 kW. So, when VD2 = 180 V, the dc-dc converter has to operate under 5.6 kW. When VD2 = 260 V, the losses in the IGBTs exceed the temperature limit at PD > 8.6 kW. So, when VD2 = 260 V, the dc-dc converter has to operate under 8.7 kW.

6. PEAK CURRENT IN THE INDUCTORS

The ferrite cores in this inductors will be magnetized magnetically saturated if the current I1 exceeds 61 A because the magnetic flux density reaches 0.3 T as calculated by (15). The dc-dc converter has to be operated under the limitation on the peak value of I1, or I1peak. The peak current imposes limitations on the dc voltage VD2. When VD1 > VD2, the peak current I1peak equals I11. When VD1 < VD2, the peak current I1peak equals I12.

Fig. 10 shows maximum transferable power when the peak current I1peak is limited under 60 A. Solid dots “*” indicate the operating points of the waveforms shown below. When VD2 = 180 V, I1peak exceeds 60 A at PD = 5.1 kW. When VD2 = 260 V, I1peak exceeds 60 A at PD = 9.4 kW.

As has been stated above, both the power loss and the peak current impose limitations on the power transfer PD and the dc voltage VD2.

Operation of the dc-dc converter has to satisfy both limitations.

Fig. 11 shows the experimental waveforms when one dc voltage is 320 V while the other is 360 V at PD = 10 kW from bridge 1 to 2. Fig. 12 shows another example of the experimental waveforms which was taken when VD1 = 320 V while VD2 = 180 V at PD = 5 kW from bridge 2 to 1. Under a set of dc voltages of VD1 = 320 V and VD2 = 180 V, the power transfer PD was limited below 5 kW as shown in Fig. 10. The peak current I1peak was 60 A in Fig. 12, as calculated in this section.

7. APPLICATION TO AN REGENERATIVE ENERGY STORAGE SYSTEM WITH GALVANIC ISOLATION

A. A 210-V, 11-kW, 2.7-kJ experimental Model

Fig. 13 shows the experimental energy storage system rated at 210 V, 11 kW, and 2.7 kJ. Circuit parameters in the dc-dc converter were the same as those in Table I. An electrolytic capacitor bank CES of 60,000 μF was used in the experiment to simulate an EDLC bank. The voltage across the capacitor bank is charged up to 350 V and discharged down to 191 V. Thus, an energy of 2.7 kJ is stored into, and released out of, the capacitor bank. It corresponds to 70% of the energy stored in CES at 350 V. The three-phase PWM converter regulates VD1 at 322 V. Its PWM carrier frequency was 11 kHz.

B. Charging and Discharging of the Capacitor Bank

Fig. 14 shows the experimental waveforms when the energy storage capacitor bank C was repetitively charged up to 351 V, and then discharged down to 191 V. The waveform of ID2, a low-pass filter with a cut-off frequency of 810 Hz. The maximal power transfer was 9.35 kW. In this experiment, the phase shift δ had a square waveform with an amplitude of 30° to make the controller in the experimental system simple. In actual energy storage systems, however,
the power transfer $PD$ should be determined by power demand, or a higher-level controller regulating the voltage on the utility grid.

7. SATURING METHODOLOGY

To start the system, if bridge 1 produced a square voltage of $v_1$ at $vD_1 = 321$ V and $vD_2 = 0$ V, a current will be flow into the inductors $L_d$ and the transformer. This current will be result in magnetic saturation in the cores of $L_d$, leading to an even larger inrush current. So, a this operation mode called “precharging operation” is presented and described to charge the capacitor bank from zero to the rated voltage of 321 V.

Fig. 15 shows the starting-up and pre charging transient waveforms of the dc-dc converter when the capacitor bank began being charged from zero to 321 V.

Here $V_{dc1}$ has been already charged up to 321 V before the DC-DC converter started the pre charging mode. Bridge 1 produced a voltage $v_1$ with a duty ratio of $d=21\%$ beside 1000% while bridge 2 was operated as a diode rectifier with all the IGBTs kept off. The first pulse of $V_1$ had a half duty ratio of $d/2 = 10.5\%$ to suppress dc magnetization of the transformer. The peak current at the first pulse of $v_1$, $I_{pp}$ is expressed as

$$ I_{pp} = \frac{\pi V_{D1} d}{\omega L} \frac{a}{2}. \quad (15) $$

In Fig. 15, the negative peak current reached 30 A, while its theoretical value was 19.8 A.

Fig. 16 shows the experimental waveforms of the dc voltage, current, and power. The starting time will be eight seconds to charge the capacitor bank from zero to 321 V. No excessive eddy current flowed into the capacitor bank.

When $vD_2$ reached 278 V, the dc-dc converter changed its operation mode from the precharging operation to the normal operation with the square voltages of $v_1$ and $v_2$ having no phase shift ($d = 100\%$ and $\delta = 0^\circ$). The power transfer $PD$ in the normal operation at $\delta = 0^\circ$ has a negative-feedback effect to balance the two dc voltages $vD_1$ and $vD_2$ due to the existence of the dead time $T_d$. Thus, after $vD_2$ reached 275 V, it was naturally charged up to the same voltage as $vD_1$.

9. CONCLUSIONS

This paper presents a bi-directional isolated dc-dc converter suitable for a regenerative energy storage system. Theoretical calculations of power losses and peak current have been verified, the dc-voltage limitations in the energy storage system. Experimental results shows that the dc-dc converter can charge and discharge the capacitor bank properly.
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Abstract: The paper is based on the hybrid approach for detecting the edges. Edge Detection plays an important role in research and technology. The novel hybrid method based on the Canny Algorithm and Neural Network. Image pre processing is done before applying the neural network. Image Smoothing performed using Gaussian Filter and Fuzzy Technique applied to convert gray level image in binary image. Finally, neural network is designed and trained for detecting actual edges. Neural network is a wonderful tool to work on real world data. It is a non-linear network with built-in thresholding capability. A classification technique is used to classify edges in an image. An Error Back-Propagation is basic technique of neural network to perform classification.

Keywords: Edge Detection, Neural Network, Canny Algorithm, Gaussian Filter

1. INTRODUCTION

Image Processing defined as the process to alter the digital image to enhance the image quality, perform filtering and classification. Areas come under image processing are image enhancement, image compression, image segmentation and edge detection. Image enhancement refers to the process of improve the brightness, sharpness of image and remove noise. In Image Compression, image is altered to reduce the size of image. Image Segmentation detects the image objects. Edge Detection refers to the process to extract the important features of image.

1.1 Edge Detection

It is an important area of image processing. The purpose is to filter out the important data from image and reduces the useless data. Edge is an important characteristic of image that characterizes the boundary of various image objects. Edge Detection is the process to identifying sharp discontinuities of image as change in intensity, depth, brightness etc. Edge Detection plays an important role in medical sciences to detect cancer etc. and in geographical sciences to analyze various areas [2]. Edges of an image can be detected by using following methods:

1.1.1 First Order Image Gradient

First Order Image Gradient method detects the edges by calculating the change in intensity of current pixel with its neighbour pixel values. List of operators comes under First Order Image Gradient like Sobel Operator, Robert Operator and Prewitt Operator [2].

1.1.2 Second Order Image Gradient

Second Order Image Gradient with laplacian finds the zero crossing means when the change in image is sharp, the derivative is zero [3].

1.1.3 Canny Algorithm

Canny Algorithm is a multistep process to detect the edges. Steps are given below [16]:

1. Canny edge detector uses the Gaussian Filter to reduce the noise in image and produces a blur image.

2. The edge can be in any direction horizontally, vertically or diagonally, so the edge detector operator returns the first derivative in horizontal direction (Gx) and vertical direction (Gy). Edge direction is identified by

\[ Q = \arctan \left( \frac{G_y}{G_x} \right) \]

\[ G = \sqrt{G_x^2 + G_y^2} \]

3. From the given values of image gradient, the direction of edge is calculated by comparing the gradient value with its local maxima. This step is also called as non-maximum suppression because it gives a wide range of edges including thin edges.

4. Once the gradient values have been computed, threshold is performed. The total number of edge points depends on the value of threshold. Large the value of threshold produce small number of edges. Small the value of threshold produce large number of edges.
5. After applying the threshold, edge thinning is performed to remove the false edges that are shown in image. It removes all the unwanted edge pixels.

1.1.4 Fuzzy Techniques

Fuzzy Techniques are used to detect the image. These techniques only work on binary image. Divide the image pixels in three categories based on intensity and after that some if then rules are used to detect edges [16].

1.1.5 Neural Network technique

Neural Network technique detects edges using two ways classification and clustering. Error Back Propagation is commonly used method comes under classification to detect the edges. Clustering is also one of the techniques to detect the edges [16].

1.1.6 Hybrid Methods

Hybrid Methods of Neuro Fuzzy are used to detect image edges based on gray level image is converted into binary and after the ANFIS is used to train the network [16].

1.2 Classification

Classification is the process to classify the data in one or more categories. Classification plays an important role to detect the edges of an image. In Edge Detection, classification is used to divide the image in two classes of edge or non-edge regions. Classification can perform through Hard Computing and Soft Computing. Both are explained below:

1.2.1 Hard Computing

Hard Computing is a conventional Computing Technique uses analytical approach to solve the problems. Sequence of steps is given to solve the problem. Input given to the system must be exact without any type of noise data. Based on the input given it produces correct result. Various Hard Computing techniques of classification are given below:

1.2.1.1 Decision Tree Classification

Decision Tree Classification forms the tree structure during the classification of pattern. Each node (except leaf node) of tree represents a question, branch represents the answer to the question and leaf node represents the class of pattern. When an input pattern is passed to the classifier, at each node question about the value of an attribute is asked and depend on the answers branches are created [15]. Questions or tests create the branches in three ways. First, by asking discrete questions. For example, pixel value equal to 100, 200, 300 etc. Second, asking continuous questions. For example, pixel value > 100 or pixel value < 100. Third, asking discrete binary questions. For example, pixel value belongs to x category gives answer yes or no. The leaf node represents the class of the pattern. The main advantage of tree based classification is to code these test cases are simple but some points must be taken care. First, the test cases must be correct, wrong questions can leads to incorrect solution. Second, the number levels in tree should be small; otherwise it can affect the speed of classifier.

1.2.1.2 Rule Based Classification

Rule Based Classification Technique classifies the pattern based on if-then rules. Rules in the classifier are defined as condition->y, where conditions check the values of attributes and depend on the condition output y comes. The output is class of the given pattern [15]. A Rule r in classification performs the conjunction on attribute values of pattern and depends on these values class of the pattern comes. This type of classifier is easy to create but the attributes chosen for conjunction must be correct because wrong conjunction can leads to incorrect classification. Any type of noise present in input data can affect the solution.

1.2.1.3 Bayesian Classification

Bayesian Classification is based on Baye's Theorem a probability based model. The model describes the probability with which pattern X with certain attribute values belongs to class C. It depends on three main factors. One, Prior probability that any pattern belongs to class C. Second, Prior probability of X, means X having satisfied attribute values required to be the member of class C. Third, posterior probability of X that says X belongs to class C [15]. Means

\[ P(C|X) = \frac{P(X|C)P(C)}{P(X)} \]  

[15]

1.2.2 Soft Computing
Soft Computing technique is based on working of human brain. It having the capability to work in the environment of imprecision and uncertainty. It is fault tolerant and easily works on noise data. As compared to Hard Computing technique that works on crisp data, it can work on fuzzy data, neural nets. Soft Computing is based on approximation means as compared to giving optimum results, it give good results. There are various techniques come under soft computing are discussed below:

### 1.2.2.1 Fuzzy Based Classification

Fuzzy Set Theory was introduced by Zadeh in 1965 to work on imprecise and uncertain data. Fuzzy plays an important role to classify this data. Classification is performed on fuzzy data set not on crisp data set. Fuzzy theory uses a membership function to classify the data. Membership function defines up to which extent the pattern belongs to a class. The applications of Fuzzy Logic image processing, decision making, pattern recognition, quality of service etc. All the aspects of imprecise and uncertainty are present in these applications. Image having lots of noise and impreciseness in it. Fuzzy can easily work on these types of applications and provide approximate solutions.

Fuzzy Logic System is automated System that performs the classification using three steps. First, perform fuzzification that converts the crisp data set into fuzzy set. Second, actual classification is performed. In third step, defuzzification is performed that converts the fuzzy data set again into crisp set.

![Fig 1.1 Structure of fuzzy image processing](image)

### 1.2.2.2 Neural Network

Artificial Neural Networks Architecture is based on human body nervous system. Neurons in brain have a very complex structure where each neuron is connected to almost $10^4$ other neurons [13]. Neurons pass energy as input to other neurons after processing. A neuron processes all the inputs at one time and produces the output. Artificial Neural Network use this feature to solve complex real world problems. Artificial Neural Network can easily work on imprecise and uncertain data. Fault Tolerance is one of the important features of neural network.

Classification and Recognition are common areas of Neural Network. Neural Network performs classification based on the training. Network is trained through supervised and unsupervised learning. Supervised learning is known as classification and unsupervised learning is known as clustering. In supervised learning, some patterns with their desired results are stored in network. Patterns are represented as n-dimensional feature space. The values for all features of a pattern are known. To classify a pattern, an artificial neural network processes each feature value at one time and produces the output.

A finite and correct set of patterns must be provided for good results. Inputs or feature values of a pattern
are denoted as \(x_1, x_2, x_3, \ldots, x_n\). For each input value a weight is given \(w_1, w_2, w_3, \ldots, w_n\).

Output = \(\sum (w_i x_i) + b\)

At each Neuron, output is calculated by summation of input*weight and bias value is added with summation denoted by \(b\). The output value defines the class of pattern. If the output class does not match with desired result, the weight value is modified based on some rule. The output is modified by two ways. In Incremental Training, the weight values are updated depend on output of each pattern but in Batch Training, after observing the output of each pattern the weight is modified. And new weight is applied to each pattern again. Each cycle is called an epoch. The output is updated until data is not classified properly. In unsupervised learning, the patterns are given without desired result. Depend on the output clusters are formed means pattern with similar properties having almost same output value.

1.2.2.3 Genetic Algorithm

Genetic Algorithm approach performs learning and refines the classification by creating new populations. The patterns or concepts in GA are also known as population. The patterns are used to classify the data. Sometimes, these patterns are not enough or strong to classify the data properly. So, from the given set of patterns, features of two or more patterns are combined to form new pattern or offspring. The patterns chosen must be suitable so the new patterns must be better than the older ones [14].

2. PROPOSED HYBRID APPROACH FOR EDGE DETECTION

The Proposed Architecture of Edge Detection based on Canny’s Algorithm, Fuzzy Technique and Neural Network. Canny Algorithm is popular for giving good results as giving sharp edges. Fuzzy Technique is popular for creating sets based on membership values of element in set. Neural is a powerful tool to work in real world, faulty and noisy data. The complete Architecture is described below:

2.1 User Interface

User Interface is a GUI through the user passes the input image and gets the output image from the Edge Detection System. An RGB image is passed as input to the system and resulting edged image is get as an output.

2.2 Image Smoothing

Canny’ Gaussian Filter applied to the input image to perform the image smoothing. Smoothing is performed to remove the noise from the image. In image smoothing, the original image is blurred with a given ratio specified by the value of \(\sigma\). 2D linear filter is given as

\[
G(x, y) = \frac{1}{2\pi\sigma^2} e^{-\frac{x^2 + y^2}{2\sigma^2}}
\]  

[17]

2.3 Gray Scale Image

Gray Scale Image is produced by applying the Gray Scale Filter. The image is converted to gray level because to convert an image into binary through gray level is easier than coloured RGB image.
2.4 Binary Image

An image defined as binary image if image having only two colors black and white. Image is converted to black and white or binary through fuzzy logic technique. The gray scale values between 0 and 80 are converted into black color and gray scale values between 80 and 255 are converted into black color.

2.5 Neural Network Error Back-Propagation Algorithm

Neural Network’s Error Back-Propagation Algorithm applied to classify the edges of the image. Network is trained with 16 patterns [8]. For each given pattern the desired output is specified. The Network is trained by applying these patterns with learning rate of 0.2.

2.6 Edge Detection

The trained Network is applied on the binary image. Image was processed four pixels by four pixels [8]. The complete image is scanned and the trained network is applied to detect the final edges.

3. RESULT

In the proposed method an input image is presented. On input image gaussian was applied to blur the image. On blurred image gray scale filter was applied to convert image in gray level. On gray level image fuzzy technique is applied to convert this image in binary image. Finally, the neural network is trained and the resulting weight is applied to the image for detecting the edges. The result of the proposed method is given below:
4. CONCLUSION

In this paper a novel edge detection approach was specified. A number of experiments are conducted on the given technique and the results are very much clear and correct. This method also gives good results for noisy image also. These results conclude that:

The implemented technique works better in contrast and intensity variations.
The lines are smooth, straight and curved wherever required.

REFERENCES


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Abstract - Pseudoknot is an important RNA secondary structure motif for structural discovery. Predicting the RNA secondary structure helps to understand its function. Earlier RNA secondary structure prediction algorithms ignored Pseudoknots because of their complex structure. But now Pseudoknots are in consideration as they enhance the accuracy of RNA secondary structure prediction. Arbitrary Pseudoknot prediction through Dynamic Programming has been proved to be NP-hard. So, now-a-days researchers are focusing on newer techniques. One such technique is the use of Formal grammar for the said prediction. This article focuses on Formal grammar algorithms for RNA secondary structure prediction including Pseudoknots.

Keywords— Pseudoknot, RNA secondary structure, Formal Grammar.

I. INTRODUCTION

Bioinformatics is the application of information technology for solving biological problems. RNA has been a focal point of researchers since a decade. The primary structure of RNA is a sequence of nucleotides (A, G, C and U) that are linked together by phosphodiester bonds. This primary structure folds upon itself to form secondary structure. The secondary structure of RNA is fashioned with Watson-Crick base pairs (G-C or A-U) and Wobble base pairs (G-U). Deriving the secondary structures of RNA remains an important research topic in RNA informatics. Several motifs have been defined in the secondary structure like helix, hairpin loop, bulge loop, internal loop, exterior loop, Multi branch loop and Pseudoknot as shown in Fig. 1[1].

A helix is formed by a sequence of base pairs. A hairpin loop is shaped when at least three unpaired nucleotides exist in a helix. An unpaired nucleotide on either side of a helix is called bulge loop. In the same way unpaired nucleotides on both sides of helix lead to formation of internal loop. An exterior loop contains the end of sequence. A multi-branch loop or junction loop is formed when various structures emerge from a helix. All the above mentioned structures carve up a common property that there is no crossing of base pairs in the helix. Contrary to it, a Pseudoknot occurs when base pairs cross making the structure complex. For this reason early RNA secondary structure prediction algorithms ignored Pseudoknots.

II. RNA PSEUDOKNOT

A Pseudoknot in RNA secondary structure is formed when two secondary structures overlap whirling it into a complex structure. If i, j, k, l represent nucleotides of a sequence such that i.k and j.l are two base pairs, then the following sequence of nucleotides will represent a Pseudoknot:

(i) \( i<j<k<l \)
(ii) \( k<j<i<l \)
(iii) \( i<l<k<j \)

Pseudoknots are not true topological knots. Pseudoknot was first recognized in the turnip yellow mosaic virus in 1982 [2]. Pseudoknots have various functional roles like ribosomal frame-shifting, translation and telomerase activity [3]. It also plays a role in aging.

III. FORMAL GRAMMAR ALGORITHMS FOR PSEUDOKNOT PREDICTION

RNA Secondary structure prediction has been in focus of researchers since 1980’s. But early RNA secondary structure prediction algorithms ignored Pseudoknots because of its
complex structure. Researchers, Rivas and Eddy in the year 1999 laid the foundation of RNA secondary structure prediction using Pseudoknots by the concept of gap matrices [4]. They further extended this concept by formulating grammar for Pseudoknots [5]. Although, gap matrices has been a good way to understand the structure of Pseudoknots, but Rivas and Eddy could handle structures up to 2 gap matrices only. So this algorithm could not handle complex Pseudoknots.

In the year 2003, Parallel communicating grammar system (PCGS) was introduced [6]. It consists of more than 1 Chomsky grammars one of which is the master grammar. Grammars for various RNA motifs have been introduced here which are controlled by master PCGS. This grammar system is then converted into stochastic version by applying probabilities on grammar rules. These probabilities have been calculated with the help of CYK algorithm [7].

Another attempt was made by [8]. They extended the Pair hidden markov models on tree structures [9] to include Pseudoknots. Their concept is based on structural alignment which incorporated tree adjoining grammar (TAG). They have defined two subclasses of TAG- simple linear TAG (SLTAG) and extended simple linear TAG (ESLTAG). These subclasses are realized with initial trees and adjunct trees. RNA substructures can be represented with the help of these trees by applying various operations like adjoin, derivation etc. Finally, tree representations are aligned to predict Pseudoknots.

Tree adjoining grammar has enough computational capacity to model Pseudoknots it takes too much time for parsing. [10]. So, researchers have tried to lower the time complexity by using other forms of grammar.

Context free grammar (CFG) has been in focus for RNA secondary structure prediction including Pseudoknots. A pattern matching technique using stochastic context free grammar was proposed by [11]. Similar to stochastic version of PCGS, probabilities are assigned to rules of the said CFG model to convert it to stochastic context free (SCFG) model. Probabilities are calculated by extending covariance model [12]. Their concept divides the secondary structure into various substructures which they call as related residues and nonrelated residues. Similar to PCGS, CYK algorithm and inside-outside algorithms are implemented to find most probable secondary structure including Pseudoknots.

Another flavor of context free grammar is multiple context free grammar (MCFG) [13]. MCFG is an extension to CFG to include functions such as bifurcation, base pairing, unpaired based etc. Each rule in MCFG is accompanied with two types of probabilities, transition probability and emission probability. Most probable tree is found by using CYK algorithm [7]. CYK algorithm is implemented using 5-dimensional dynamic programming matrix. MCFG is converted into SMCFG version with the help of inside-outside algorithm [7]. As CYK algorithm takes too much space, so hashing is used to reduce the space complexity.

Another attempt to model Pseudoknots was done by [14] using context sensitive grammar (CSG). Contrary to the context free grammar, context sensitive grammar considers the context in which it is situated. Because of this, it is easy to model various RNA structures. But unfortunately there is no known automatic parser for CSG. [15] has tried to parse the said CSG by augmented transition network. They haven’t explained how their model handles Pseudoknots explicitly.

TAGRES, a subclass of TAG was proposed by [10] to model Pseudoknots. Initially, the time complexity of TAGRES was O(n^4). So, they constructed stochastic regular grammar (SREG) to filter the candidate structures, thereby reducing the time complexity. Stochastic parameters are calculated using logarithm of probability values. For testing the efficiency of system, two types of grammars are generated, one with optimized probability parameters and the other with un-optimized probability parameters.

Path controlled grammars were also used to model Pseudoknots [16]. Path controlled grammars extend context free grammars to restrict their path from root to leaf. Tree structures are built keeping a check on the restriction imparted on grammar. This way the same grammar can be controlled to generate various RNA structures including Pseudoknots.

IV. CONCLUSION AND FUTURE SCOPE

Various Grammars have been used by researchers to model RNA secondary structures including Pseudoknots. But till date automatic RNA Pseudoknot prediction remains an open challenge. Researchers restrict the type of structures in order to achieve lower space complexity and time complexity. Moreover, Pseudoknots exist in various forms. Automatic prediction of various types of Pseudoknots is another open problem. Our future work will be to predict arbitrary Pseudoknots.

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Semi-hidden target recognition in gated viewer images
Fused with thermal IR Images

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Abstract: in many areas such as Defense and security supervision scenarios typically deals with detection and classification of suspects in complex and dynamic backgrounds. Imaging systems is introduced for this purpose should therefore provide imagery that produces optimal simultaneous recognition of both suspects and their context. In this review the recognition of semi-hidden targets, which are targets that are enclosed in complex scenes, and which may either be combined or merged with other details in the scene. Imagery of semi-hidden targets obtained with conventional visual (Cameras) and Infra-a Red (IR) camera is typically not optimal for recognition and classification purposes. Previous studies on image fusion did not consider semi-hidden targets. This study reflects the potential benefits of adding a laser range gated viewer (GV) to an IR camera and of fusing GV and IR imagery for the recognition of semi-hidden targets. A combination of an Image Quality Metric (IQM) and an accurate saliency metric is used to select a fusion method that is optimal for semi-hidden target recognition. In the end, results of both metrics are validated through a human observer experiment. For such application where very complex scenes (in which target recognition remains difficult after fusion) a background dimming algorithm is used that either uniformly dims the entire background or applies less dimming in the local target background or in regions with important contextual information, without affecting the target representation itself. The optimal combination of fusion method and amount of dimming is determined through a second observer experiment. In a third observer experiment, we tested if target motion influences the preferred amount of dimming. It is found that fusing GV with IR imagery improves human recognition of semi-hidden targets. A simple pixel-based approach with a PCA-based weighted fusion scheme appears to be the optimal fusion method. Contextual dimming improves target recognition in complex backgrounds. In addition, moving objects appear to affect observer’s dimming preference.

Index Term: - Gated viewer, IR images, IR thermal Images, Fusion Algorithm, Dimming algorithm, Image Quality metric.

I. INTRODUCTION

Visual target detection and recognition play a key role in military operations and security operations. Correct recognition is required in order to make the correct decision within juridical and ethical bond or (in the most extreme cases) to minimize insured damage. Defense and security applications involve time-critical counter in complex and dynamic, urban and rural situations where objects easily molded in with their surroundings and where it is hard to distinguish between militant and safe targets. Therefore it is important to understand the context of a situation. Throughout this study the term object recognition applies to object classification in combination with context understanding.

In conventional visual Detector like cameras or CCD footage (TV) and infrared (IR) images of complex scenes targets are sometimes hard to distinguish when they are masked or semi-hidden (i.e. partly occluded by or merged with other objects in the scene). To resolve this problem we propose to extend conventional system or IR systems with a laser range gated viewer (GV). A GV laser system uses the time of flight of a laser pulse to set a range gate around the target, thereby achieving an optimal target representation in combination with background suppression. Gated Viewing systems make use of a time controlled camera and a pulsed laser illumination source. A short laser pulse illuminates the scene. Because the camera timing is locked to a specific distance (with nanosecond accuracy), the shutter only accepts light from this given distance, and the system only represents the object of interest (within a small depth window, here 15 or 45 m). Since both distance and depth are controlled by camera timing, it is possible to
selectively image an object at a certain distance with a prescribed depth range. Objects outside the prescribed time (and hence depth) window will not be represented in the image, and will therefore not disturb the observation. In this way, one can also avoid nearby atmospheric scattering sources in the optical path. As a result, GV images are highly suitable for human or automatic classification of semi-hidden targets. The fusion of GV images with conventional systems or IR images may enable optimal target classification (provided by the GV images) without loss of context (provided by the conventional TV or IR images). To achieve this goal the fusion algorithm should optimally represent both the salient target details (edges, lines, points, corners etc.) which are important for object fixation and recognition, and the background details which are relevant for context recognition. Since conclusion may help humans to better understand the object, it is important to preserve this information as well.

The goal of this study is to explore if human semi-hidden target recognition can be improved by fusing GV and IR imagery. To the best of image fusion and the benefits of laser range gated viewing for semi-hidden target recognition have not been in before. In this study, both a simple weighted averaging fusion scheme and fusion using a multi resolution image representation. Fused imagery implemented for human inspection should provide sufficient spectral and spatial details. In practice, this implies that object boundaries (i.e. edges, corners, curves, lines and points) should be clearly represented, with high contrast and a minimal amount of noise. Multi-resolution image decompositions are typically well suited for this purpose.

In this an Image Quality Metric (IQM), validated by a human observer experiment, to determine the optimal fusion method. The advantage of human observer validation is that selection of the fusion method can be tuned to a specific scenario such as the one considered in this study, since observers know what information is relevant for a specific task. For the IQM, the Piella and Heijmans Image Quality Metric (which is based on the metric of Wang and Bovik as it both accounts for human vision and is especially designed for the evaluation of fused images (because it focuses on important details transferred from both in-put images into the fused image). To the best of our knowledge previous studies did not deploy a human observer validated IQM to select the optimal fusion method for a given task.

In some scenarios the target turns out to be perfectly blended in the background (i.e. target and background have nearly the same intensity even after optimal fusion). For these scenarios a dimming algorithm is developed and an adapted quality metric to achieve optimal background dimming. Again the results are validated with a human observer experiment. Based on the results, an approach for multi sensor semi-hidden target recognition using GV and IR images can be described and it is possible to assess whether GV improves the human recognition task of semi-hidden targets.

Gated viewing is the decisive technology to see under the worst environmental conditions. The technology is not well known as it is military related and they want to keep this out of the public interest. This technology can easiest be explained that strobe a short laser pulse in its transmission at speed This pulse and what is illuminated is viewed by an image intensifier where the shutter is controlled with nano sec. accuracy. The shutter can be opened and controlled precisely. This makes it possible to decide whether you want to see and how far away. The depth of the illuminated zone can be controlled. By this technology you can control where you want to see and obstacles out side this area are black. For example, driving a car in heavy snow fall. Lights give reflexes making you close to blind. By using this, it can illuminate behind the snow and see objects as negatives which are very comfortable. Aqua Lynx is optimized for submerged applications. Sea Lynx is optimized for marine search and rescue, Nord Lynx is for airborne search and rescue and low altitude flying. These are useful in medical areas.

Semi-hidden images

Semi-Hidden images are the set of objects which are molded into surroundings, which makes it difficult to identify whether it is harmful or just a scene. For example:
The dataset used in this study was recorded by the Swedish Defense Research Agency (FOI). It consists of LWIR (Long Wave IR: 8–12 lm) and GV image sequences (1.55 lm) representing seven scenarios in the same rural background. In each scenario a human (either in a standing or kneeling position) is located at a different position in the scene at approximately 2 km distance, resulting in different levels of contrast with the vegetation. The IR images are 768 X 578. This example of the GV and IR images of a typical scenario. In this it is assumed that target detection has been performed by e.g. change detection, movement detection, LIDAR or RADAR, and that both images have been registered.

We distinguish 3 different regions in the scenes used in this study:

- Region I or Target region: the target support,
- Region II or Target background region: the local Target background excluding the target support, and
- Region III or Scene background region: the area represented only in the IR image.

The target region contains the object to be classified (in this case a human) and is represented in the GV image. The target background region is represented by both the GV and the IR images. The scene background region is only represented by the IR image.

II. METHODOLOGY

[1] Fusion Algorithm

The similar amount of information in the GV and IR images is different for each of the three image regions as defined in previous images. As a result different fusion methods may be applied in each of these regions. In the background region the scene is only depicted through the IR image and therefore it simply extract in the fusion process. In the Target region and the Target background region the scene is represented in both image aspects. The GV image provides satisfactory approach for target representation in the Target region. In this study it is assumed that the images have been registered and are in pixel-wise order. Now it is consider both pixel based fusion methods and region based fusion methods.

To apply these two type: pixel based and region based approach, we used averaging approach which can be classified into two i.e. global weighting and local weighting. Both on the entire image (pixel based) or within a region based

Additional weighted averaging fusion algorithms are also investigated: three multi-resolution decomposition fusion schemes. Transform methods based on image decompositions that represent both directional high frequency components and image approximating low frequency components enable simultaneous maximization of salient details and background preservation. Moreover, another three methods that are known to accurately represent directional information. The three selected methods are: Dual-tree complex wavelet transforms (DT-CWT), Curvelet transform and Contourlet transform.

In order to limit the amount of fusion results, and because it is not our goal to establish the optimal settings of each specific representation method, the input parameters of the modules (e.g. number of scales/levels) are set to values that are known.

For weighted fusion, four different fusions rules are applied. The simplest rule is averaging by a weight of 0.5. The advantage of this method is its simplicity; however, a huge disadvantage is the risk of reducing the contrast of important details.

Therefore also consider more advanced methods for defining the weights. Principle Component Analysis (PCA) to define a single weight for the entire image. The eigenvector belonging to highest Eigen value of the covariance matrix represents the weights. Since PCA gives greater weight to the image with more energy, this may result in giving preference to one image and ignoring the other image due to a significant difference in variance. Because of the disadvantages of the above two rules, two additional rules are considered that define weights per pixel.

A method using the local energy is defining the weight: by comparing the local variance in a window
for each pixel and the other by comparing the local maximum intensity (the sum) of the window. The advantage of these rules is that important information is considered pixel wise. However, a disadvantage is that it may result in a dotted pattern in the fused image. With the help of four rules of low frequency component averaging is ensured.

Low frequency component an averaging rule ensures that the approximations of both input images are covered. So other fusion rules are:- Local-maximum, Local-variance and Maximum-local-variance.

For the high frequency components with guiding information, the maximum selection rule is in selective manner a good fusion rule as it preserves the important hidden details that are captured in these components. When there are unwanted features in one image type at a specific location, these will be preserved as well this is a drawback of this approach. An example is an edge that is not precisely represented in one image type due to poor resolution whereas the other image type represents the edge of the same object more accurately. Both edges will be preserved with the maximum-selection rule, may result a bad edge effect in the fused image. Because of this disadvantage we also considered a selection of other fusion rules: Average, Maximum-average, Maximum-local-variance and Local-maximum-selection.


After applying fusion algorithm, it becomes difficult to recognise the target as it approximates the intensity of target with background. For such images background dimming algorithm provides a better improvement and recognition and therefore results in classification of target with background in comparative manner. On the other side of this technology it also may degrade the context recognition. This shows that there should be a degree at which background dimming should be applied so that features could be preserved.

Dimming algorithm is done through three types of methods: First of all global dimming, which is dimming of the entire background by using a dim factor in between 0 and 1 while the target remains unchanged. A dim factor closer to 0 means more dimming. The second method, or local dimming, the dim factor is only applied to the boundary of the target; the boundary being the difference between the representations of a binary object and its dilation. For both dimming methods the dimming equation is given by:

$$F_d = d^* kF + (U - k)F$$  \[1\]

in which $F$ is the fused image, $d$ is the dim factor, $k$ a binary mask to indicate the area to be dimmed, $U$ an array of size $F$ consisting of ones and $F_d$ the dimmed image. The binary mask $k$ contains ones over the entire background area when global dimming should be applied or ones at the location of the target boundary and zeros elsewhere when local dimming should be applied.

There is also a third dimming method that includes a saliency metric that indicates background areas containing important context information. This metric defines priority regions in an image related to human object fixation. It is important that the saliency metric is related to the operator task. A map that defines and high-lights these regions is called a saliency map. The saliency map is used to restrict dimming to background areas that are relatively unimportant for the given task. The map has values from 0 to 1: it reaches a value of 1 for the important areas and zero else where. In this way the background is dimmed with only little loss of context. We call this contextual dimming equation (1) then become:

$$F_d = (d + (1 - d) Sal)^* kF + (U - k) F$$  \[2\]

in which $Sal$ is a saliency map with the same size as $F$. For $Sal$ values of 0 the dim factor is used to dim and elsewhere less dimming applies up to no-dimming for $Sal$ values of 1.

There are two ways to create a task specific saliency map: top down and bottom up. Top-down saliency maps are more complex and more accurate representations of task, specific saliency as they highlight the areas that contain important objects with respect to the task. Top-down saliency maps can be achieved either by prioritizing objects with respect to the observer’s task and training a classifier to perform the task, or by human fixation studies. However, these top-down methods are very time consuming and complex.

Therefore we considered simpler bottom-up methods. Bottom-up methods map human priorities by highlighting areas of interest using methods that
approximate fixation points or visual attention. The simplest and least performing saliency metric is variance. However, variance does not predict object quality by humans. Moreover, variance is sensitive to noise. Therefore there are three more advanced methods based on availability and results of comparison with human conspicuity tests.

[1] Frequency Tuned Saliency or FTS.
[2] Harris Points of Interest or simply Harris.

[4] Image Quality metric

To select the optimal method to fuse image for human recognition humans have to evaluate a large number of participating features. When such type of situation comes then it takes quite large time for processing. Therefore an Image Quality Metric algorithm can be used for reducing human evaluation work for fusion method.

Wang and Bovik[12] developed a universal objective quality metric, or image quality index(Q0), which models the amount of distortion or common information between original image and a processed image. It uses a correlation coefficient, a luminance factor and contrast measure in which \( l_i \) is the mean, \( r_i^2 \) the variance, \( r_{ij} \) the covariance and \( I_1 \) and \( I_2 \) the two input images. It has a value between -1 and 1, 1 being the best quality. It cannot used for evaluation purpose at the end of the procedure.

Paella and Heijimans[14] adapted a different Image Quality Metric by combining \( Q(I_1,F) \) with \( Q(I_2,F) \) where \( I_1 \) and \( I_2 \) are two input images and \( F \) is the fused image Where \( w \) being the window of the set of windows \( W \). \( k_w \) is a weight based on saliency in window \( w \). Typically the local variance is taken.

However, a disadvantage is the lack of a standard saliency metric. Therefore, the saliency map can be directly substituted in the IQM for the variance; thus \( k = \text{Sal}_1/\text{Sal}_1 + \text{Sal}_2 \). Thus both the methods are used.

The IQM is used to enable the selection of the optimal amount of dimming. In general, dimming the background decreases the entire quality of the image whereas the quality of the target representation increases. By multiplying the quality of background with the quality of the target area for each dim factor, a curve is obtained from which an optimum can be seen. The background quality is defined by the amount of IR context available in the image. Therefore the basic Wang and Bovik algorithm for IR with \( F_d \) is used to quantify the background quality. With this metric more dimming results in lower quality. Therefore, another metric is chosen to measure the target quality: target-background contrast. A factor is added to the quality of the target, which functions as a tuning parameter in order to fit the optimum to results of observer experiments. The dimming quality metric (DQM) is then defined as image quality with a dim factor has mean intensity of the target, and mean intensity of background and \( a \) is a parameter used to fit this function according to the observer.

III. EXPERIMENTAL PROCEDURE

In this section all step by step experiments are formed starting from Fusion algorithm. As it is seen earlier that IQM and DQM are used to select best fusion algorithm.

These methods are first validated by the user after that these fusion algorithm are applied. Moving objects are also considered by using dimming algorithm. This experiment procedure is divided into three parts:

1. Fusion algorithm selection.
3. Tests regarding the influence of moving objects.

In this part results of various sections are shown:
1. Fusion results

The six results shown are part of the selected subset of eight results in step 2. The fusion algorithms corresponding to the subset of step 2 which are selected for the human ranking experiment in step 3a are:

1. Pixel based contourlet fusion with average fusion rule (weight of 0.5) for low frequency components and max–min-selection fusion rule for high frequency components, or PC5. [2]
2. Pixel based weighted average fusion with a weight of 0.5, or PW5. [1]
3. Pixel based weighted average with weights defined using PCA, or PWP. [5]
4. Pixel based contourlet fusion with local-maximum fusion rule for low frequency components and maximum-selection fusion rule for high frequency components, or PCM. [1]
5. Region based Priority fusion, or RPF. [1]
6. Region based contourlet with local-variance fusion rule for low frequency components and maximum-selection fusion rule for high frequency components, or RCV.
7. Region based contourlet with average fusion rule (weight of 0.5) for low frequency components and maximum-selection fusion rule for high frequency components, or RC5. [1]
8. Region based weighted average with weights per pixel defined by local-maximum, or RWM.

Fig. 3. Examples of patches (regions I and II) for fusion results top to bottom left to right IR, GV (as provided by FOI), 1 (PC5), 3 (PWP), 4 (PCM), 5 (RPF), 8 (RWM), 7 (RC5). The numbers and acronyms are corresponding to the human.

Results of step 7 shows average observer ranking of the best pixel based fusion methods, for all 7 scenarios (denoted by 1–7). Fusion method based on pixel based, Weighted Average/PCA gives a average ranking of 2.86 which is the first in rest of the scenario.

Results of step 7 shows average observer ranking of the best region based fusion methods, for all 7 scenarios (denoted by 1–7). Fusion method based on region based, showed that curvelet/ maximum-local-variance, maximum-selection; curvelet/ maximum-local-variance, local-maximum-selection; curvelet/ maximum-local-variance, maximum-local-variance and Priority Fused all ranked(Average ranking) first in all the scenarios.

Results can be seen in these set of pictures.
7 for pixel based and region based fusion, top initial IR image, bottom left pixel based weighted fusion with weights defined with PCA and bottom right region based priority fusion.

2. Dimming results

![Undimmed and dimmed with three types of dimming and dim factor = 0.4. Clockwise from top left: undimmed, global dimming, local dimming and contextual dimming.](image)

![Table for dimming factor for Pixel based.](image)

![Table for dimming factor for Region based.](image)

PC & RC: - pixel based contextual dimming and Region based contextual dimming respectively. 
PG & RG: - pixel based global dimming and Region based global dimming respectively. 
PL & RL: - pixel based local dimming and Region based local dimming respectively.

3. Dimming In a video- stream

The observer does this step as mentioned in step 11, as it review the parameters that affect the image at static or moving object. It would be clearly be explained by using these results.

![Dim Factor](image)

IV. Conclusions and Discussions

In this review it is seen that semi -hidden target recognition can be improved with fusion of GV image and IR imagery. The main purpose of GV is as achieved in this review as for complex environment; objects are being identified without loss of context information to some extent.

In this review global and context dimming algorithm is concluded. With pixel based approach and with PCA based approach weighted fusion because GV represent only a small area of the IR image and only contains single objects. In IQM of pixel based it showed that multi-resolution methods that provides good results. To get the details of local background, it suggested region based approach, but observer find that negative on complex image create a good impression on target in pixel based approach.

In future work it might be useful to design and test a region based fusion approach in which the background region is fused rather than simply selecting IR. To get best results participation of observer should increased as per environment.

While in case of moving object in a video-stream it appears to affect the observer preferences of using dimming factor. However it only affects more on static objects than on moving objects.

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Review: Face Recognition Using Principal Component Analysis

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Abstract: A facial recognition system is a computer application for automatically identifying or verifying a person from a digital image or a video frame from a video source. Human often uses faces to recognise persons. Face Recognition is one of the most important Bio-metric method. Biometric can be used for authentication purpose because Authentication plays a very important role in various applications related to security. Biometrics refer to automatic recognition of individuals based on their physiological and behavioral characteristics. Person identification is one of the most crucial building blocks for smart interactions. Among the person identification methods, face recognition is known to be the most natural ones, since it is the technique we use to identify people in our everyday lives. In this paper we will talk about a problem that is very complex and common, namely face recognition. Identifying and comparing faces in images is a very complex task, this is probably why it has attracted so many researchers in the latest years. We will also describe PCA (Principal Component Analysis) algorithm that’s one of the widely used algorithm for face recognition. We will also describe the advantages of OpenCV, a freeware library for implementing face recognition. Various techniques used in face recognition will also be covered and also future of the Face Recognition will also be covered.

INTRODUCTION

Research on automatic face recognition systems have been conducted now for almost 60 years. The first paper talking about face recognition can be traced back to the 1950’s in psychology [3]. The first work concerning automatic face recognition was done in 1970 by Kelly [7]. His thesis describes a computer program which performs a complex image processing task. The task was to find the same person in a set of images taken by a TV camera.

During the 1980’s, work on face recognition had no progress. But the interest grew rapidly again from the beginning of the 1990’s. [9] gives some reasons why the research interest increased: real-time hardware became more available, and the importance of surveillance-related applications increased.

Different Approaches

In the beginning of the 1970’s, face recognition was treated as a 2D pattern recognition problem [7, 2, 6]. The distances between important points were used to recognize known faces. E.g. measuring the distance between the eyes or other important points or measuring different angles of facial components. But the main focus has been on making face recognition systems fully automatic.

Feature extraction methods

Feature extraction is the task where we locate facial features, e.g. the eyes, the nose, and the chins etc. This tasks may be performed after the face detection task. Or they may be performed at the same time, but that is if the face detection task locates features in order to find out where the face actually is inside the image.

A big challenge for feature extraction methods is feature “restoration”, this is when the system tries to recover features that are invisible due to large variations, e.g. Head pose when we are comparing a frontal image with a profile image.

Principal Component Analysis:

PCA is a useful technique that has found applications in many fields like Face Recognition and Image Compression and is a common technique for finding patterns in data of high dimensions. PCA is an unsupervised method. By using Principal Component Analysis, patterns are detected in given data and similarities and dissimilarities are identified based on these detected patterns.

PCA, Which makes the use of Eigen faces, is the technique pioneered by Kirby and Sirivich in 1988. With PCA, the probe and gallery images must be the same size and must first be normalized to line up the eyes and mouth of the subjects within the images. The PCA approach is then used to reduce the
dimension of the data by means of data compression basics and reveals the most effective low dimensional structure of facial patterns. This reduction in dimensions removes information that is not useful and precisely decomposes the face structure into orthogonal (uncorrelated) components known as eigen faces. Each face image may be represented as a weighted sum (feature vector) of the eigen faces, which are stored in a 1D array. A probe image is compared against a gallery image by measuring the distance between their respective feature vectors. The PCA approach typically requires the full frontal face to be presented each time; otherwise the image results in poor performance. The primary advantage of this technique is that it can reduce the data needed to identify the individual to 1/1000th of the data presented. PCA involves following:-

1. Computing the Eigen Values and Eigenvectors from original data.
2. Deciding which Eigenvectors are significant and form a Feature Vector.
3. The Eigenvector with largest Eigen Values is termed as Principal Component of data set.
4. Higher Eigen Value has more significant Eigenvector.
5. Forming a new coordinate system based on Feature Vector.
6. Mapping the data to new space.
7. Reduce Complexity of data by reducing its dimensionality.

PCA is the technique which selects the features of the image which vary the most from the rest of the image. Each eigen face represents only a certain feature of the face.

OpenCV: Free Source Library used for face Recognition

OpenCV (Open Source Computer Vision Library) is a library of programming functions mainly aimed at real time computer vision, developed by Intel. It is free for use. The library is cross-platform. It focuses mainly on real-time image processing. The OpenCV Library is a collection of low-overhead, high-performance operations performed on images. The OpenCV implements a wide variety of tools for image interpretation. It is compatible with Intel® Image Processing Library (IPL) that implements low-level operations on digital images. In spite of primitives such as linearization, filtering, image statistics, pyramids, OpenCV is mostly a high-level library implementing algorithms for calibration techniques (Camera Calibration), feature detection
(Feature) and tracking (Optical Flow), shape analysis (Geometry, Contour Processing), motion. Analysis (Motion Templates, Estimators), 3D reconstruction (View Morphing), object segmentation and recognition (Histogram, Embedded Hidden Markov Models, Eigen Objects). The essential feature of the library along with functionality and quality is performance. OpenCV has so many capabilities it can seem overwhelming at first. Some of them are as follows:

(i) General computer-vision and image-processing algorithms
    (mid-and low-level APIs).

(ii) High-level computer-vision modules.

(iii) AI and machine-learning methods.

(iv) Image sampling and view transformations.

(v) Methods for creating and analyzing binary images.

(vi) Methods for computing 3D information.

(vii) Math routines for image processing, computer vision, and image interpretation.

(viii) GUI methods.

ADVANTAGES OF OPENCV OVER MATLAB FOR FACE DETECTION

(i) Ease of Use

OpenCV is more easy to use than MatLab. OpenCV is more cross-platform and it can run on almost all the systems irrespective of OS while MatLab is also cross-platform but it has some issues while running on apple and Linux based computers.

(ii) Speed

Matlab is a pretty high-level scripting language. Matlab is built on Java, and Java is built upon C. So when we run a Matlab program, your computer is busy trying to interpret all that Matlab code. Then it turns it into Java, and then finally executes the code. OpenCV is basically a library of functions written in C. We are closer to directly provide machine language code to the computer to get executed. So ultimately we get more image processing done for our computers processing cycles, and not more interpreting. As a result of this, programs written in OpenCV run much faster than similar programs written in Matlab.

(iii) Resources Needed

Due to the high level nature of Matlab, it is a resource hog. Matlab code requires over a gig of RAM to run through video. In comparison, a typical OpenCV program only require ~70mb of RAM to run in real-time. Similarly, a full installation of Matlab plus all the toolboxes will take up a couple of gigs on computer. In comparison, OpenCV only requires a gig at most.

(iv) Cost

List price for the base (no toolboxes) Mat-Lab (commercial, single user license) is around 1500 euro while OpenCV is free for use.

(v) Development Environment

Matlab comes with its own development environment. So when we use Matlab, we have to use its programming environment and IDE as well. For OpenCV, there is no particular IDE that we have to use. Instead we have a choice of any C programming IDE depending upon whether we are using Windows, Linux or OS X.

(vi) Memory Management

OpenCV is based on C. As such, every time you allocate a chunk of memory you will have to release it again. While due to the high-level nature of Matlab, it is “smart” enough to automatically allocate and release memory in the background.

(vii) Portability

Matlab is having the feature of portability but it causes some problems while running on Linux and apple OS while OpenCV has no such issues. OpenCV can run without any issues on any platform be it Windows, Linux or OS X.
Based on some other criteria, we can conclude that OpenCV is much better than MatLab.

4. The Future of Face Recognition

Since the beginning of year 2000 there have been produced numerous papers concerning face recognition. The website www.face-rec.org holds a list over recently released work and also a list with links to different research groups.

The future of face recognition systems looks bright, there is over 50 groups spread around the world working on the issue today.

5. CONCLUSION

In this paper we have looked at three different approaches on how to design a face recognition system. The first approach was the feature extraction method. This method is widely used to create individual vectors for each person in a system, the vectors are matched when an input image is being recognized.

An intelligent system should have an ability to adopt over time. Reasoning about images in face space provides a means to learn and subsequently recognize new faces in unsupervised manner. When an image is sufficiently close to face space (i.e. it is face like), but it is not classified as one of the familiar faces, it is initially labeled as “unknown”. The computer stores the pattern vector and the corresponding unknown image. If a collection of “unknown” pattern vectors cluster in the pattern vectors cluster in the pattern space, the presence of a new but unidentified face is postulated.

The Eigen face approach to face recognition was motivated by information theory, leading to the idea of basing face recognition on a small set of image features that best approximate the set of known face images, without requiring that they correspond to our intuitive notions of facial parts and features.

REFERENCES

Inverse Computation of Microwave Scattering Signals for Foreign Object Identification Such as Breast Screening

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Abstract: Many of the Engineering, Computer Science and Mathematical problems require adopting proper inverse computation methods to find solutions. This paper presents one of the novel inverse algorithms which can be used for detecting unknown objects by processing data obtained from a few microwave measurements. Complex backscattered electromagnetic waves characterise the relations of the internal properties of the host material. By applying a microwave signal from the surface of the host and measuring both the inward and scattered waves, it is possible to extract certain critical information of the internal structure. The theoretical solutions based on the breast model have given promising results when tested experimentally. In the case of experimenting for breast tumour detection, this research builds a prototype breast model which has similar electrical properties of both normal and malignant breast tissues and then obtains measurements with and without a tumor like inclusions. With the measured data alongside the theoretical data, an efficient computer program can be developed to identify and then compute the size and the location of the tumour. This outcome would provide an enormous support for further developing of this microwave detection technique before proceeding to clinical trials. This method also can be developed for the industry for quality measurements, engineering automation and applications, moisture and property measurements and for several in-line applications. Each application needs specific research and development approach to build the microwave applicator and, with the algorithms developed in two-dimensional and three-dimensional approaches, it is possible to obtain expected outcome for the use of health sector and many industry applications.

Keywords—Breast screening, Microwave detection, breast tumor, inverse problem

I. INTRODUCTION

Solving many physical problems require proper data acquisition methodology using either a well-prepared system model or an appropriate experimental measuring system. The system that the problem exists can be modeled in a mathematical environment to identify and construct the appropriate forward problem. The inverse problem of the system based on the forward problem has to be indentified first and then develop and construct the inverse algorithm for computing expected solutions. In case of foreign object detection, the inverse algorithm is used to calculate the important critical parameters of the targeted objective inside the host material. In this experimental research paper we discuss microwave data processing algorithms for both forward and inverse problems and present the methodology for obtaining solutions based on an iterative approach.

Microwaves can be used as a safe and non-invasive method for foreign object identification. Forward and backscattered signals are used to calculate the impedance and reflection coefficients as a function of the applied microwave frequency for early stage breast cancer detection. Screening mammography is the most effective method used for breast cancer detection but it suffers from a number of drawbacks such as; high false-positive and false-negative rates, a possible risk factor, discomfort for the examinee and their difficulty in tolerating breast compression [1, 2]. In our approach the microwave signal applied from the surface of the breast skin requires a minimal compression of the breast for accurate measurements.

Earlier research carried out for internal property measurement of dairy products and food samples have shown that microwave imaging is feasible using a dielectric permittivity profile obtained from a suitable measuring system [3, 4]. Further, ex-vivo measurements taken using the Keam Holden VE2 analyser [5] have shown that a tumour has a significant difference in complex dielectric permittivity to that of healthy breast tissue. Studies carried out by Harness et al. [6, 7] illustrate the opportunities for active microwave sensing in the
breast. X. Li et al. [8, 9] have developed a time-domain approach using ultra wideband radar technique to investigate the presence and the location of malignant breast tumours.

Our approach analyses the behaviour of the microwave signal and computes the distance from the surface of the host material. The type of signal used in this application is a uniform plane wave which penetrates through the non-homogeneous internal structure. In practice, there are losses due to finite conductivity and lossy dielectric but these are usually very small and can be neglected [10, 11]. The behaviour of the signal with different material properties have led to general equations which can be obtained from the well-known theory of electromagnetic wave propagation [12, 11]. Those equations contain information on the electrical and magnetic properties of the internal structure and can be used to develop algorithms to compute the unknown parameters of the internal object. The front-end microwave measurement provides us with the required information which is needed to identify and then to compute these parameters.

II. METHODOLOGY

The basic model for the microwave measurement system is shown in Figure 1. The measurement system provides the microwave signal to the antenna system which sends the radio signal into the host material. The backscattered signal from the internal structure of the host is received by the same antenna system and sends it back to the measurement system for analysis.

The approach here is to consider the internal object as a “wave-scatter” and solve the two-dimensional inverse problem in cylindrical coordinates to compute the unknowns. The two-dimensional study lays the groundwork for the practical computation of the inverse method using a simple microwave measurement system. Although the forward problem is simple the inverse problem in a two-dimensional application which may have practical complications due to the complexity of multiple scattering [13], multipath and diffraction effects in non-homogeneous internal structure of the host material. In order to avoid any incompatibility we treat both forward and inverse problems separately, so that our inverse algorithm will be viable for the computation of unknowns based on measured data. The front-end antenna system receives the complex backscattered signal to provide the required data for our inverse algorithm. In both transmitted and received signals, amplitude and phase changes are expected and will be measured a number of times at different frequencies.

A. Forward Problem

Microwave signal applied to the host from the antenna aperture is scattered at the tumour boundary. Therefore, the boundary condition between tumour and non-tumour cells is the significant wave scattering condition for achieving successful results. This situation is modeled in Figure 2 for constructing the forward equation of this problem.
Fig. 2. Modeling the breast for the microwave application. (a) The two-dimensional breast model. (b) Experimental implementation.

Fig. 2(a) shows three antennas in the xy-plane, each of which can transmit and receive microwave signals at the surface of the breast skin. We assume the internal structure of the breast is homogenous [14] and loss-free for the microwave signal. The cylinder inside the model represents a tumour with radius \( a \) (here presumed to be perfectly conducting, for simplicity).

Fig. 2(b) shows the two-dimensional coordinate system of the breast model with a single antenna. The wave excited from the antenna is z-polarized and travelling in the \( x \)-direction towards the wall. We consider two situations with this arrangement; one is scattering from the wall by itself and the other is scattering from both the wall and the cylinder.

Assume the wave to be finite at point \( C \) (the centre of the cylinder) and periodic with a period \( 2\pi \) in the polar angle. For plane wave illumination, the reflected wave at the point \( O \) due to the cylinder [15, 16] is

\[
E_{r,\phi}^r(p, \phi; f_i) = E_0 \sum_{n=\infty}^{\infty} j^{-an} \left[ -\frac{J_n(k_i a)}{H_n^{(2)}(k_i a)} \cdot H_0^{(2)}(k_i p) \right] e^{j\omega t} \tag{1}
\]

MERGEFORMAT

where \( f_i \) is the frequency of the transmitted signal, \( a \) is the radius of the cylinder and \( p \) is the distance to \( O \) from the centre of the cylinder. Here, \( j = \sqrt{-1} \) and \( k_i \) is the wave number of the medium given as \( k_i = 2\pi f_i \sqrt{\mu \epsilon} \) where \( f_i \) is the frequency of the microwave signal, and a time dependence \( e^{j\omega t} \) is assumed, where \( \omega_i = 2\pi f_i \). Also, \( \mu \) and \( \epsilon \) are the permeability and permittivity of the medium, respectively. The factors \( J_n(\cdot) \) and \( H_n^{(2)}(\cdot) \) are the Bessel function of the first kind and the Hankel function of the second kind, respectively [16]. The subscript \( i \) relates to the frequency \( f_i \) of the signal used.

The reflected wave received at the antenna is captured at its aperture and sent back to the measuring instrument. In this xy-plane, the scattering about the circular boundary of the cylinder is not uniform but rather it depends upon the incident angle. Suppose we rotate the point \( O \) to \( O' \) through an angle \( \pi - \phi \). Then it will lie along the \( x \)-axis and the distance \( p \) becomes \( d_i \) which is the distance to the centre of the cylinder from the antenna \( A_i \). Now, the field at the point \( O' \) can be found as (using equation (1) with \( \phi = \pi \)).

\[
E_{r,\phi}^r(f_i) = E_0 \sum_{n=\infty}^{\infty} j^{-an} \left[ -\frac{J_n(k_i a)}{H_n^{(2)}(k_i a)} \cdot H_0^{(2)}(k_i d_i) \right] \tag{2}
\]

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The total field at the antenna plane is the sum of all the field components along its vertical line. Using equation (1), the average field captured by the xy-plane of the antenna aperture (within the angle \( 2\theta \)), can be found by adding together the entire field values across the antenna and normalising (averaging) over the aperture of angular width \( 2\theta \), as

\[
E_{r,\phi}^r(f_i) = 2E_0 \sum_{n=\infty}^{\infty} j^{-an} \left[ -\frac{J_n(k_i a)}{2\theta H_n^{(2)}(k_i a)} \right] \int_{\theta/2}^{\pi/2} H_0^{(2)} \left( \frac{k_i d_i}{\cos(\theta)} \right) e^{j\omega t} d\phi \tag{3}
\]

MERGEFORMAT

where \( \theta = \tan^{-1}\left( B / 2d_i \right) \) is the angle that \( d_i \) makes with the \( x \)-axis when \( O \) is rotated to lie at either end of the antenna aperture (Fig. 1b). The numerical value of the argument of the Hankel function in equation (2) varies with the position of the object: the distance from the centre of the cylinder to the xy-plane of the antenna aperture is the only variable in the argument. This distance varies from \( d_i \) to \( d_i \sec \theta \) within the aperture plane. Therefore, in order to obtain a reasonable solution for the length \( p = d_i \sec \phi \) we replace the argument of the Hankel function in equation with \( k_i d_i / \cos(\theta / 2) \) by taking the average
value of $\phi$ over the interval $(\pi - \theta, \pi)$. The reflected wave is expressed in cylindrical coordinates but the parameter $d_l$ needs to be modified to become $d_l^* = d_l + \Delta$, where $\Delta$ is a parameter with units of length which accounts for the curvature of the wavefronts in the aperture of the horn antenna. Here, we have values of $d_l^*$ in far zone fields $\Delta$ may be equated to unity (the paraxial approximation). One can consider further details on the estimation of $\Delta$ are in Senaratne [17]. Then, we have

$$E_{z}(r) = 2E_{0} \sum_{n = 1}^{\infty} \left[ -\frac{J_n(k(a))}{20H_{n}^{(2)}(k,a)} H_{n}^{(2)} \left( \frac{k d_{l}^{*}}{\cos(\theta/2)} \right) \right] \int_{-\pi}^{\pi} e^{jin} d\phi.$$ 

When the wave is radiating from large apertures and the distance to the object is comparatively large, one can consider $\theta$ to be very small so that $\cos(\theta/2)$ may be equated to unity (the paraxial approximation in far zone fields [18]). For the practical application here, we have values of $d_l$ somewhere between the near-field and the far-field zones of the antenna radiation.

The negative, positive and zero terms in equation are separated to obtain

$$E_{z}^{-}(f) = 2E_{0} \sum_{n = 1}^{\infty} \left[ -\frac{J_{n}(k(a))}{20H_{n}^{(2)}(k,a)} H_{n}^{(2)} \left( \frac{k d_{l}^{*}}{\cos(\theta/2)} \right) \right] \int_{-\pi}^{\pi} e^{in} d\phi$$

$$E_{z}^{0}(f) = 2E_{0} \sum_{n = 1}^{\infty} \left[ -\frac{J_{n}(k(a))}{20H_{n}^{(2)}(k,a)} H_{n}^{(2)} \left( \frac{k d_{l}^{*}}{\cos(\theta/2)} \right) \right] \int_{-\pi}^{\pi} e^{in} d\phi$$

$$E_{z}^{+}(f) = 2E_{0} \sum_{n = 1}^{\infty} \left[ -\frac{J_{n}(k(a))}{20H_{n}^{(2)}(k,a)} H_{n}^{(2)} \left( \frac{k d_{l}^{*}}{\cos(\theta/2)} \right) \right] \int_{-\pi}^{\pi} e^{in} d\phi.$$ 

Equation is simplified to yield

$$E_{z}^{-}(f) =$$

$$E_{z}^{0}(f) =$$

$$E_{z}^{+}(f) =$$

From the Bessel function relationships we have

$$J_{-n}(z) = (-1)^{n}J_{n}(z)$$

$$Y_{-n}(z) = (-1)^{n}Y_{n}(z)$$

$$H_{n}^{(2)}(z) = J_{n}(z) - jY_{n}(z),$$

and we simplify the first term of the equation (6) to obtain

$$2E_{0} \sum_{n = 1}^{\infty} \left[ -\frac{J_{n}(k(a))}{20H_{n}^{(2)}(k,a)} H_{n}^{(2)} \left( \frac{k d_{l}^{*}}{\cos(\theta/2)} \right) \right] \left( 1 - e^{in} \right) =$$

$$2E_{0} \sum_{n = 1}^{\infty} \left[ -\frac{J_{n}(k(a))}{20H_{n}^{(2)}(k,a)} H_{n}^{(2)} \left( \frac{k d_{l}^{*}}{\cos(\theta/2)} \right) \right] \left( 1 - e^{in} \right).$$

Then the averaged solution for the reflected wave in equation (6) becomes

$$E_{z}^{-}(f) = E_{0} \left[ -\frac{J_{n}(k(a))}{20H_{n}^{(2)}(k,a)} H_{n}^{(2)} \left( \frac{k d_{l}^{*}}{\cos(\theta/2)} \right) \right] +$$

$$2E_{0} \sum_{n = 1}^{\infty} \left[ -\frac{J_{n}(k(a))}{20H_{n}^{(2)}(k,a)} H_{n}^{(2)} \left( \frac{k d_{l}^{*}}{\cos(\theta/2)} \right) \right] \int_{-\pi}^{\pi} e^{in} d\phi.$$ 

The expression in (9) is, however, not quite the final result. There is an additional scale factor that needs to be applied to account for the fact that the electric field distribution in the antenna aperture is not uniform due to the TE$_{01}$ mode propagating in the waveguide. The aperture field has the following form:

$$E_{z}(y, z) = E_{0z} \cos \left( \frac{\pi y}{B} \right)$$

where in (10), $E_{0z}$ is the maximum field strength in the aperture. It should be noted that for the case we have here, when the cylinder diameter is much smaller than the aperture dimension $B$, and with the cylinder axis located at $y = 0$, the incident field on the cylinder has an amplitude approximately equal to the maximum field strength in the aperture ($= E_{0z}$). It is therefore virtually unaffected by the non-uniformity of the aperture field. In receive mode, however, the non-uniform aperture field must be accounted for in some way.

The procedure now is to introduce an equivalent uniform field distribution spanning the entire antenna aperture. The field amplitude of this equivalent distribution is the average aperture field $E_{z}$, which is evaluated as follows:
In (11), the average field amplitude can be seen to be a factor \(2/\pi\) times the maximum field amplitude. The average sensitivity of the antenna to incoming radiation from the cylinder is therefore reduced by the factor \(2/\pi\) compared to the case of a uniform field distribution in the aperture. This gives the final result for the average scattered field as follows:

\[
E_{\text{scat}}(\mathbf{r}) = \frac{2E_0}{\pi} \left[ -\frac{J_n(kr)}{H_0^{(2)}(kr)} \right] + 2\sum_{n=1}^{\infty} \frac{J_n(kr)}{H_n^{(2)}(kr)} \sin\left(n\theta/\pi\right) \left(\frac{\sin(n\theta)}{n\theta}\right),
\]

(12)

where the variable \(\mathbf{r}\) in (12) is the average arc length to the aperture plane from the centre of the cylinder, that is,

\[
\mathbf{r} = d_1 \frac{\cos(\theta/2)}{\cos(\theta/2)}.
\]

Consider now the transmitted plane waves from the antenna. In this experiment, we normalized the analyser to zero decibels (1 milliwatt power) and zero phase at the antenna front face. Since the cylinder is small compared to the aperture width we have

\[
E' = E_0 e^{ikd_1}.
\]

(13)

The reflection coefficient is defined as the ratio of the reflected signal to the incident signal at the load [16, 17, 18]. Therefore the reflection coefficient of the cylinder measured at the antenna front face relative to the centre of the cylinder is

\[
\Gamma = \frac{-E_{\text{scat}}(\mathbf{r})}{E_0 e^{ikd_1}} = \frac{2}{\pi} e^{-ikd_1} \left[ -\frac{J_n(kr)}{H_0^{(2)}(kr)} \right] H_0^{(2)}(kr) - 2\sum_{n=1}^{\infty} \frac{J_n(kr)}{H_n^{(2)}(kr)} H_n^{(2)}(kr) \sin\left(n\theta/\pi\right)\left(\frac{\sin(n\theta)}{n\theta}\right).
\]

(14)

In (14), the minus sign in the definition of \(\mathbf{r}\) stems from the normalisation to a short circuit at the aperture plane.

**B. Measurements and Data Acquisition**

The backscattered field measured at a single antenna is now studied in detail. The outwards signal from the antenna was directed towards a blank concrete wall. A copper circular cylinder, which represents the “object” in this experiment, was placed in front of the antenna parallel to its front face. An electromagnetic horn antenna was used to match the waves from the guiding system to a large radiating aperture. The design parameters of the antenna flange follow the basic concepts of the wave guide theory [19]. The bore sight axis of the antenna is perpendicular to the wall. The wave excited by the antenna is z-polarized and travels in the \(x\) direction towards the wall in the \(xy\)-plane (Fig. 2(b)). The complex reflection coefficient of the wall measured at \(O'\) can be written as

\[
U_w e^{i\theta_w} = \Gamma_w e^{i2k(d_1+\ell)}
\]

(15)

where \(\Gamma_w\) is the magnitude of the complex valued reflection coefficient of the wall. Our interest is to find \(U_c e^{i\theta_c}\) the complex reflection coefficient of the conducting cylinder at \(O'\) which has a distance \(d_1\) from the centre of the cylinder,

\[
U_c e^{i\theta_c} = \Gamma_c e^{i\theta_c}
\]

(16)

where \(\Gamma_c\) is the magnitude of the complex valued reflection coefficient of the cylinder.

Considering the separate measurements that can be obtained only with the wall and therefore, from the bilinear transformation [20] we have
\[
U_c e^{j \theta_c} = \frac{d U_{c+w} e^{j \theta_{c+w}} + h}{d U_{c+w} e^{j \theta_{c+w}} + 1} \tag{17}
\]
where \( U_{c+w} \) is the magnitude of the complex reflection coefficient measured at \( O' \) with both the wall and the cylinder present and \( \theta_{c+w} \) is the corresponding phase angle. In the experiment, we measured \( U_{c+w}, \theta_{c+w}, U_w \) and \( \theta_w \) with several frequencies. We used these data in equation (17) to find \( U_c \) and \( \theta_c \). Good approximate results are observed by setting \( g = 1, q = 0 \) and \( h = -U_w e^{j \theta_w} \) and so, the complex reflection coefficient of the cylinder can be found as

\[
U_c e^{j \theta_c} \approx U_{c+w} e^{j \theta_{c+w}} - U_w e^{j \theta_w}. \tag{18}
\]
In equation (18) we have also included the terms involving the phase angle in the simplification of equation (14).

C. Measurement System

Here we reproduce the experimental results obtained from microwave measurement system as explained in [20]. We used a network analyser to analyse the amplitude and phase properties in this experiment. This instrument provides the microwave signal to the antenna system that sends the radio signal into the host material. The backscattered signal from the internal structure of the host is received by the same antenna system and is sent back to the analyser for measurement. A coaxial waveguide was used to send and receive microwave signals between the source instrument and the antenna system. The width of the aperture is \( B = 0.308 \) m and the length (into the page in Fig 1(b)) is \( A = 0.235 \) m.

The measured data at the network analyser is then processed using the reconstruction algorithms. An electromagnetic horn antenna was used to match the waves from the guiding system to a large radiating aperture. The size of the waveguide feed has been chosen so that the desired operating frequency of 2-3 GHz lies within the frequency range for which only the transverse electric (TE_{01}) mode propagates. This fixes the cut-off frequency, according to the fundamental relationship

\[
f_c = \frac{c}{\lambda_c}, \tag{19}
\]
where \( \lambda_c \) is the cut-off wavelength and \( c \) is the velocity of light (approximately \( 3 \times 10^8 \) m s\(^{-1}\)). Therefore, when frequencies of 2-3 GHz are used, this system operates well above the cut-off.

It is not possible to measure the reflection coefficient of the cylinder directly from this experiment. Instead we computed these using two sets of measured data with and without the cylinder. The cylinder is moved away from the antenna in 5 mm steps to obtain measurements. For each position of the cylinder, the measurements are repeated twenty times at selected frequencies. The results at each frequency are averaged.

We measured \( U_{c+w} \) and \( \theta_{c+w} \) with both the wall and the cylinder, and \( U_w \) and \( \theta_w \) with only the wall present, the cylinder having been removed. The values of the reflection coefficients of the cylinder can be calculated from these measurements using equation (18).

Fig. 3 shows both calculated and measured reflection coefficients of cylinder plotted against the distance \( d_1 \). Theoretical calculations were carried out using equation (14) and measured results were obtained using the experimental microwave data acquisition system according to equation (18). Here we find, in general, the calculated and measured data are very similar.

D. Inverse Problem in the Two-dimensional Study

In the inverse problem the field quantity at the receiver is known by measurement and \( a \) and \( d_1 \) are the unknowns. Finding solutions to two unknowns is not possible unless we obtain two equations by using two different frequencies. Use of this method for an over-determined system is also possible, however, the result may be degraded unless suitable frequencies are used [21]. (That is, they need to be different and have wavelengths of the same order as the unknown cylinder size \( a \).) At the antenna point we record amplitude and phase measurements for a range of frequencies. We use two equations for two different
frequencies to find \( a \) and \( d \). Then, the inverse solutions are found using Newton’s iterative method [22]. The solution of the above system often needs several iterations and the number of these mainly depends upon the number of unknowns and the value of the initial guess.

Now we form the general equation using the measurement values and the forward equation as found from equation (14). The general equation has two constituent equations of the form

\[
\Delta \Gamma = \begin{bmatrix}
\Delta \Gamma_{1,1}(a, d_1) \\
\Delta \Gamma_{1,2}(a, d_1)
\end{bmatrix} = \begin{bmatrix}
a \\
d
\end{bmatrix}
\]

(20)

where we now use the subscripts 1,1 and 1,2 in \( \Gamma \) to indicate two different frequencies. In full,

\[
\Delta \Gamma = \begin{bmatrix}
\Delta \Gamma_{1,1} \\
\Delta \Gamma_{1,2}
\end{bmatrix} = \begin{bmatrix}
\Gamma_{1,2} - \Gamma_{1,1} \\
\Gamma_{1,2} - \Gamma_{1,1}
\end{bmatrix} = \begin{bmatrix}
a \\
d
\end{bmatrix}
\]

(21)

where \( \Gamma_{1,1} \) and \( \Gamma_{1,2} \) are the field measurements from the antenna for two different frequencies. Solution of the inverse problem can be used to find the size and the location of the internal object using the reflection coefficient measurements.

We tested the computed results for the solution of equation (18) using two sets of guess values. We used the measurement results obtained with the 1.3 cm diameter cylinder. In the iteration processes it shows that the estimates of the two unknowns \( a \) and \( d \) converge towards their exact values (\( a = 0.0065 \) and \( d = 0.0365 \) metres). Each set of guess values needs a different number of iterations. In general more iteration is required when the guess values are further away from the exact values. The method of computing unknowns using the inverse method has been explained in details in our previous papers [15, 21]. It is emphasised that although we used known values of \( a \) and \( d \), to calibrate the forward wave, in the inverse method they are determined from the measurements by being unknown in both the forward and backward wave expansions. Thus we have a criterion to measure the success of the method.

### III. MICROWAVE SCANNING RESULTS

Using the above method we calculated the cylinder radius \( a \) and the distance \( d \) for the measured values of the complex reflection coefficients. The measured values of the magnitude and phase of \( \Gamma_{1,1} \) were used in equation (18) to proceed with the subsequent calculations in the inverse method [20]. Each time the scanning process stops when the stopping criteria is met (when \( | \Delta \Gamma_{1,1} | \) is a minimum). Simulations were carried out for a selected range of 20 Grid Points (GP).

For every pair of the guess values in the scanning range we tested with 60 iterations for convergence. A fair comparison of some estimated and actual \( a \) and \( d \) values can be found in [20]. These show that there is a good agreement.

The above results are for antenna position \( A_1 \) (Fig. 1(a)). However, using the same procedure, it is possible to find the corresponding parameters for other antenna positions. Once these calculated results are available the location of the object can be easily found using simple geometry. In practical application for the detection of a breast tumour, it would be important to measure using at least three antennas. The complex reflection coefficient of the chest wall and the skin reflection can be estimated using a set of pre-recorded data or, at the time of obtaining measurements, by comparison with the measurements that can be obtained from normal tissue when the object or tumour is absent. We acknowledge that this may be difficult in practice.

In the experiment, the actual values of \( a \) and \( d \) are 0.0065 m and 0.0415 m, respectively. For comparison, in Fig. 4, we have plotted some estimated and actual \( a \) and \( d \) values. These show that there is a good agreement.
Fig. 4. Calculated and actual results of the cylinder radius \( a \) and the distance \( d \) plotted against the actual distance to the cylinder from the antenna.

**Conclusions**

This experiment has revealed that the inverse computation of measurement data obtained from a suitable experimental set-up can provide some critical information of the scattering objects inside the host which is under test. The computation robustness and accuracy depend upon the method of data acquisition and its completeness. The laboratory experiment discussed here is to test a new approach which has a potential for developing a novel breast tumour identification technique. The new method uses low power microwave signals to detect and locate a foreign object within a homogeneous medium.

In the experimental test, the measured complex reflection coefficients agree with those predicted using the forward equation of the theoretical model. Also, the computed values of the size and the distance of the object, found with the proposed inverse method, are very close to their exact values. These computed values of the size and the distance of the object, found with the proposed inverse method, are very close to their exact values. These experimental and theoretical results indicate that the reconstruction of an internal object is feasible using measured reflection coefficients with microwave frequencies. Similarly, this method can also be further developed for many industry applications such as moisture and quality measurements.

**ACKNOWLEDGEMENTS**

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Predicting success of Nursery Schools using Artificial Neural Networks
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Abstract—Predicting the success of nursery schools based on the performance of its students in selection for good primary school, board exams, different competitive exam like engineering entrance exam, Medical entrance exam as well as performance of its students in extracurricular activities and as a result ranking them has intrigued many scholars and academic industry leaders. It is a challenging problem as the nursery schools play an important role in shaping the future of students. Parents find it very difficult to choose a school for their ward as Nursery school helps in laying down a healthy foundation for the all-round development of their child. In this paper, the use of artificial neural networks in predicting the success of nursery schools is explored. We convert this prediction problem into a classification one, i.e. , we classify nursery schools based upon success of its students in board exams and different competitive exams in one of the five categories, ranging from ‘flop’ to ‘top’. Because our neural network model can predict the success of a nursery schools, it can be used as a powerful decision aid by parents for selecting a school as well as by school management to know their strong & weak points & can therefore make appropriate changes. We have chosen to use the Error Back Propagation learning algorithm and the Multi Layer Perceptron (MLP) network architecture to formulate our problem. In this paper, we have discussed various parameters used in our model. We’ve started collecting data & would come up with the results soon.

Keywords: admission, classification, prediction, Neural-Network back-propagation, Nursery-School, multilayer perceptron.

I. INTRODUCTION
Selecting a Nursery school for their ward is one of the most difficult and lifetime decision for parents as it is the building block of their child’s elementary education. It helps in laying down a healthy foundation for the all-round development of their child. It is necessary to provide a child with good pre-primary education for him to be prepared for future education. For most analysts, admission counseling is a process of premonition and wild guess, due largely to the difficulty and uncertainty associated with predicting the school demand. Such unpredictability makes the school business a risky endeavor for the investors to take in today’s highly competitive world. No one can tell us how a school is going to Despite the difficulty associated with the unpredictable nature of problem domain, several researchers have attempted to develop models for forecasting the success of a school in gaining maximum admissions in an academic year, primarily using statistics based forecasting approach. It is the first attempt to use the artificial neural network for predicting success of nursery schools.

In our study, we explore the use of neural networks in forecasting the performance of nursery schools. We convert this prediction problem into a classification one, i.e. , we classify a nursery school based upon success of its students in selection for good primary school, board exams, different competitive exams as well as extracurricular activities in one of the five categories, ranging from ‘flop’ to ‘top.

The remainder of this paper is organized as follows. Section 2 briefly reviews the basics of ANN & also presents the comparison between the statistical techniques & our ANN approach. Section 3 gives the details of our methodology by specifically talking about the data, the neural network model, the experiment methodology and the performance measures used in this study. Next we discuss the overall contribution of this study, along with its limitations and further research directions.

II. REVIEW OF ANN
A neural networks ability to perform computation is based on hope that we can reproduce some of the flexibility and power of human brain by artificial
means. Basically a neural network is a machine that is designed to model the way in which the brain performs a particular task or function of interest. The network is usually implemented by using electronic components or is simulated in software on a digital computer.

The property of neural network that is of primary significance is the ability of network to learn from its environment, and to improve its performance through learning. A neural network learns about its environment through an interactive process of adjustments applied to its synaptic weights & bias level. After completing the learning process successfully the network is ready to be deployed for independent functioning.

Neural networks have been applied to an increasing number of real world problems of considerable complexity. The most important advantage is that artificial neural networks are capable of solving problems that are too complex for conventional technologies – problems that do not have an algorithmic solution or that solution is too complex to be found.

The application domain of neural networks today touches almost the entire sphere of science. These include: Association, Clustering, Classification, Pattern Completion, Regression & Generalization, Forecasting, Optimization etc to name a few.

Many application bibliographies exist. However, none of these include an application in forecasting the performance of colleges in gaining max admission during the counseling session. This study is one of the first to attempt the use of neural networks for addressing this challenging problem that has drawn the attention of many researchers in such areas of decision support systems and management science.

We differentiate our study from the others as follows. First, there is no reported study on using neural networks to predict the college performance in counseling session. Our study seems to be the first attempt of its kind in this problem domain. Another distinguishing feature of our study comes from its longitudinal nature. Our study is based on five consecutive years of data that covers movies released between 1997 and 2007. Our study also compares the difference between those of individual years and the combined data set of all 10 years. The results suggest that our neural networks model performs better than the one ones reported in the literature.

III. METHODOLOGY & DATA & VARIABLE DEFINITION

Though commonly known as black box approach or heuristic method, in the last decade, artificial neural networks have been studied by statisticians in order to understand their prediction power from a statistical perspective (Cheng & Titterington, 1994; White, 1989a, b, 1990). These studies indicate that there are a large number of theoretical commonalities between the traditional statistical methods, such as discriminant analysis, logistic regression, and multiple linear regressions, and their counterparts in neural networks, such as multi-layered perceptron, recurrent networks, and associative memory networks.

In our study, we explore the use of neural networks in prediction. Here, we convert the prediction problem into a classification problem. Multi layer perceptron (MLP) neural network architecture is known to be a strong function approximator for prediction & classification problems. MLP is capable of learning arbitrarily complex non-linear functions to an arbitrary accuracy level. Thus it is a candidate for exploring the rather difficult problem of mapping college performance to the underlying characteristics. The error back-propagation algorithm presents the best mapping; it is thus used in this approach.

DATA & VARIABLE DEFINITIONS: In our study, n nursery schools in NCR region are being used. The variable of interest in the study is the total number of admission based on merit only without any donation. It does not include any management seat etc. We have converted this forecasting problem into a classification one i.e. a nursery school based on its total number of seats filled (on the basis of merit) is classified in one of the five categories, ranging from ‘FLOP’ to ‘TOP’. We plan to use eight different types of independent variables. Each categorical-independent variable is converted into 1-of-N binary representations. Thus we get a number of pseudo-representations that increases the independent variable count from 8 to 28.

A neural network treats these pseudo variables as different mutually exclusive information channels. All
pseudo representations of a categorical variable will be given a value of 0, except the one that holds true for the current case, which will be given the value of 1. For e.g. the variable ‘Fee Structure’ would be represented with two pseudo variables – High & Low.

We now present a brief description of the variables chosen.

1) Primary school admission: - The most common thing mostly people see before choosing any nursery school is how it prepares their wards for admission in good primary school. The better is its result for admission in good primary school, more famous a primary school would be. There are 4 possible ratings in this category, viz, High (>80%), Fair (60 – 80%), Average (40-60%) & Poor (<40%).

2) Child to Teacher ratio: - One of the important criteria to select a nursery school is the child to teacher ratio. The lower the ratio, the more individual attention is available to your child. We classify this independent variable into 4 categories- Excellent (< 6), Good (< 10), Average (10-20) & poor (> 20)

3) Location: - Is yet another very important feature that determines a school’s fate. Any nursery school located at some prime location, e.g., within a city will surely be benefited, as all parent might want to choose a school that is closer to their home. Thus we categorize Location onto 3 main categories. These ares- Within a city, NCR, Distant area.

4) Facilities provided & infrastructures: - Any parent seeking admission for their ward in a nursery school looks forward to get some basic facilities from the school. These common facilities that everyone wants to get are: (a) Transportation Facility( b) Every preschool classroom should have a block area where children can learn to construct. (c) A preschool classroom should include spaces and materials for manipulative play, problem solving, and science exploration (d) the classroom should be equipped with a CD, tape, or record player; a variety of music, including marches, folk songs, and nursery rhymes; simple musical instruments; and a place where children can sit or march in a circle. For art, there should be tables and chairs, a sink, a non carpeted floor area, and plenty of eye-level wall space where the children's work can be attractively displayed. Easels, drying racks, and whiteboards are desirable, as well. Based on these parameters we classify this independent variable in 4 classes- Good; Fair; Average & Poor.

5) Teachers: - Teachers of a school plays a very vital role in deciding the future of the school. Good & dedicated teacher ensures good academics, which in turn ensures good learning in children & therefore better all-round development of a child. The decision variable teachers may be divided into 3 main classes- Good, Fair, Average.

6) Fee structure: - This forms yet another important parameter in deciding any school’s fate. Three different situations reside within this variable.

a) The school is good in all respects & the fee is nominal, mostly in case of good govt. school, central school. Then such a college is surely going to earn maximum number of students.

b) The school is good in all respects but the fee structure is also high, in such cases people, people don’t mind taking admission, despite high fee structure because they’re getting all the facilities & features desired from a good nursery school. So the parameter high fee structure then takes a backseat.

c) The last situation is that the school is not good i.e. it does not have good infrastructure, good faculty & does not provide any other facilities also but its fees is Low. Now in such a situation despite the fact that the fees are low, people will decide against taking admission in such schools. This implies that Low fee parameter cannot save a school if it is poor in other areas. We’ve assigned 2 values to this variable. One value is ‘Low’ & the other value is ‘High’

7) Performance of students in higher classes and other competitive exam: - Nursery schools lay the foundation for the future. They not only develop the basic skills in a child, they also play important role in how child would fair in higher classes. We thus classify this independent variable into 5 classes, viz, Excellent, Good, Fair, Average & Poor.

8) Market Value: - This decision variable, represented by 3 pseudo representations – High, Medium & Low, shows the market value of the school in current scenario. It tries to depict how much the school is able to earn through its name. Data for this variable has been collected by
surveys i.e. by considering expert opinion as well as common man’s notion.

A summary of above-mentioned variables is given in Table 1. In all there are 28 decision variables & 5 output variables that we intend to use in our approach.

Many researchers in the past few years have studied the performance of neural networks in predicting a variety of classification problems over a wide range of different business settings. Many of these studies, however, were based on a single experiment, and/or the method of selecting the training and testing samples was not clear. We believe that because of

<table>
<thead>
<tr>
<th>S. No.</th>
<th>Independent Variable Name</th>
<th>No. Of Values</th>
<th>POSSIBLE VALUES</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Primary school admission</td>
<td>4</td>
<td>High (&gt;80%), Fair (60-80%), Average (40-60%), Poor (&lt;40%)</td>
</tr>
<tr>
<td>2</td>
<td>Child to teacher ratio</td>
<td>4</td>
<td>Excellent; Good; Fair; Average</td>
</tr>
<tr>
<td>3</td>
<td>Location</td>
<td>3</td>
<td>City; NCR; Distant place</td>
</tr>
<tr>
<td>4</td>
<td>Facilities Provided &amp; infrastructure</td>
<td>4</td>
<td>Good; Fair; Average; Poor</td>
</tr>
<tr>
<td>5</td>
<td>Teachers</td>
<td>3</td>
<td>Good; Fair; Average</td>
</tr>
<tr>
<td>6</td>
<td>Fee Structure</td>
<td>2</td>
<td>High; Low</td>
</tr>
<tr>
<td>7</td>
<td>Performance of students in higher classes</td>
<td>5</td>
<td>Excellent; Good; Fair; Average; Poor</td>
</tr>
<tr>
<td></td>
<td>and other competitive exam</td>
<td></td>
<td></td>
</tr>
<tr>
<td>8</td>
<td>Market Value</td>
<td>3</td>
<td>High; Medium; Low</td>
</tr>
</tbody>
</table>

The stochastic nature of the neural network training, better experimental design methods are necessary to develop the objective performance measures of neural networks.

As opposed to using a single neural network experiment to base our results upon, we chose to follow a more statistically sound experimental design methodology, called k-fold cross-validation. In k-fold cross-validation, also called rotation estimation, the complete dataset (D) is randomly split into k mutually exclusive subsets (the folds: D1, D2,..,Dk) of approximately equal size. The classification model is trained and tested k times. Each time (t ∈ {1,2,.., k}), it is trained on all but one folds (D\Dt) and tested on the remaining single fold (Dt). The cross-validation estimate of the overall accuracy is calculated as simply the average of the k individual accuracy measures. Since the cross-validation accuracy would depend on the random assignment of the individual cases into k distinct folds, a common practice is to stratify the folds themselves. In stratified k-fold cross-validation, the folds are created in a way that they contain approximately the same proportion of predictor labels as the original dataset. Empirical studies showed that stratified cross validation tend to generate comparison results with lower bias and lower variance when compared to regular k-fold cross-validation (Kohavi, 1995). In this study, to estimate the performance of neural network classifier a stratified 10-fold cross-validation approach is used. In 10-fold cross-validation, the entire data set is divided into 10 mutually exclusive subsets (or folds) with approximately the same class distribution as the original data set (stratified). Each fold is used once to test the performance of the classifier that is generated from the combined data of the remaining nine folds, leading to 10 independent performance estimates.

IV. CONCLUSION

Our approach of using artificial Neural Networks in this field aims to classify the nursery schools in one of the five output categories & thus predict their success. We’ve started implementing our problem & would come up with the results soon. This study is the first attempt to use Neural Networks for addressing this challenging problem that combines two different application domains of Predicting & Classification & brings out the much-desired output. This model would be highly beneficial to the:
a) **Anticipating Parents**- It would be very useful for parents seeking admission for their wards for they’ll have all the comparative information that is needed & therefore at a glance they can decide which school to opt for.

b) **School Management**- They would be benefited for they can come to know their strong & weak points & can therefore make appropriate changes.

Compared to the other model types (i.e. logistic regression, discriminant analysis and classification and regression trees), using the exact same experimental conditions, neural networks performed significantly better. Because the neural network models built as part of this study are designed to predict the success of a nursery school in gaining maximum admissions in an academic year before the actual admissions take place, they can be used as a powerful decision aid by parents and school managements.

From an application perspective, once developed to a production system, such a neural network model can be made available (via a web server or as an application service provider) to academic decision makers, where individual users can plug in their own academic related parameters to forecast the potential success of a school. A neural network model can be designed in a way such that it can calibrate its weights (continuous self learning) by taking into account new samples as they become available. Much additional work, in terms of modeling extensions, further experimentation for testing the performance, and applications to other media product demand forecasting, remains to be done.

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Video Image Processing: Detection and Tracking Of Color

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Abstract— Image processing has been used for many applications in last few years but mapping the address of the colors through the front camera of the Mobile phones has been not used till now. Encapsulation of the position of the image color been pointed is being done and then following the color with the motion of the image been displayed whenever it is jagged in front of secondary camera, and analysis of the image is shown. This has been done using the software Matlab in the PC.

This paper proposes that there is a possibility of the mapping of colors through the front camera of mobile phone.

I. INTRODUCTION

Image processing is any form of signal processing for which the input is an image, such as a photograph or video frame; the output of image processing may be either an image or a set of characteristics or parameters related to the image. Most image-processing techniques involve treating the image as a two-dimensional signal and applying standard signal-processing techniques to it. These techniques are applied to it using a computer. Though image processing usually refers to digital image processing, but it also has two other types namely:

1) Analog Image Processing
2) Optical Image Processing

An image is considered to be a function of two real variables, for example, a(x,y) with “a” as the amplitude (e.g. brightness) of the image at the real coordinate position (x,y). On the basis of these dimensions an image can be converted into signals and manipulated or processed accordingly as our need therefore apart from image processing there are two other image manipulations techniques as follows:

A. image analysis (image in -> measurements out)
B. image understanding (image in -> high-level description out)

<table>
<thead>
<tr>
<th>Name of the technique</th>
<th>Description</th>
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<tbody>
<tr>
<td>Image processing</td>
<td>Image in -&gt; image out</td>
</tr>
<tr>
<td>Image analysis</td>
<td>image in -&gt; measurements out</td>
</tr>
<tr>
<td>Image understanding</td>
<td>Image in -&gt; high-level description out</td>
</tr>
</tbody>
</table>

A. Purpose of Image processing: The purpose of image processing is divided into 5 groups. They are:

- Visualization - Observe the objects that are not visible.
- Image sharpening and restoration - To create a better image.
- Image retrieval - Seek for the image of interest.
- Measurement of pattern – Measures various objects in an image.
- Image Recognition – Distinguish the objects in an image.

Every image is an array of small pixels. Each pixel specifies a color and is represented on the x and y axis in a two dimensional window as shown is Fig. 1.

![Example of a grayscale pixilated image](image_url)

Fig. 1. Example of a grayscale pixilated image.
An array for the above image can be represented as follows:

```
int[ ] myArray = { {236, 189, 189, 0},
```
II. OBJECTIVE

This paper focuses on the detection of red color from an image/video input. Live image/video input is taken through the front camera of a mobile phone. The video input is in the forms of frames which are nothing but images in the forms of arrays as explained above. The video input contains a red object. This red object is moved in front of the camera. This video input is analyzed frame wise. Each frame contains the same red object but at different positions. The position of the red object is calculated on the X and Y axis in each frame and is thus recorded. Simultaneously these recorded positions are tracked and on each respective tracked position a colored dot is placed on a white canvas, at that position in accordance with the same X and Y axis criterion, inside the mobile. Thus as the red object moves, simultaneously an image is drawn, on the canvas, along the path on which the object is moving. This drawn image can thus be saved as a file with an extension ‘.jpg’, ‘.png’ or ‘.bmp’. All this process is done using the software Matlab on a computer but we want to propose the same idea for a mobile phone using java.

The software used for converting the matlab code into java is JA compiler. The compiler converts the code into a ‘.jar’ file which is a java class file.

III. SOFTWARE USED

MATLAB® is a high-level language and interactive environment for numerical computation, visualization, and programming. Using MATLAB, you can analyze data, develop algorithms, and create models and applications. The language, tools, and built-in math functions enable you to explore multiple approaches and reach a solution faster than with spreadsheets or traditional programming languages, such as C/C++ or Java™.

MATLAB JA BUILDER: Together, MATLAB, MATLAB Compiler, and MATLAB Builder JA enable you to develop applications using MATLAB, and then incorporate them in Java programs. You use MATLAB—a high-level, matrix-optimized language with built-in math, graphics, and data analysis functions—to rapidly prototype, implement, and test your algorithms. Once an application is complete, you use the builder to automatically package MATLAB code as Java classes. These classes can be integrated in a Java application and referenced in the same way as standard Java classes.

Like other Java components, the JAR files generated by the builder are generally platform-independent and run on any platform supported by MATLAB.

To create portable Java classes from MATLAB functions, MATLAB Builder JA encrypts the functions and generates a Java wrapper around them. To deploy these classes, you first install the MATLAB Compiler Runtime (MCR) on a target computer, and then run the classes incorporated in a Java program against the MCR. The MCR (provided with MATLAB Compiler) is the full set of shared libraries required for executing MATLAB based components. It provides complete support for all features of the MATLAB language and most toolboxes.

IV. METHODOLOGY

(A) DETECTING AND TRACKING THE RED COLOUR

Suppose our input video stream is handled by vidDevice object.

Step 1: First acquire an RGB Frame from vidDevice object.

MATLAB Code: rgbFrame = step(vidDevice);

![rgbFrame](image)

Step 2: Extract the Red Layer Matrix from the RGB frame.

MATLAB Code: redFrame = rgbFrame(:,:,1);
Step 3: Get the grey image of the RGB frame. MATLAB Code: 
```matlab
grayFrame = rgb2gray(rgbFrame);
```

Step 4: Subtract the grayFrame from the redFrame. MATLAB Code: 
```matlab
diffFrame = imsubtract(redFrame, grayFrame);
```

Step 5: Filter out unwanted noises using Median Filter MATLAB Code: 
```matlab
diffFrame = medfilt2(diffFrame, [3 3]);
```

Step 6: Now convert the diffFrame into corresponding Binary Image using proper threshold value. Change its value for different light conditions. Suppose in my code I have used its value as 0.15. MATLAB Code: 
```matlab
binFrame = im2bw(diffFrame, 0.15);
```

V. ALGORITHM

The blocks present in the below flowchart are being explained below in the following steps:

Step 1: Capture the video frames using the video input function.
Step 2: Set the properties of video object.
Step 3: Start the video acquisition.
Step 4: Set a loop that starts after 50 frames of acquisition.

This loop contains the following steps:
1. Get the snapshot of the current frame.
2. Now to track the red objects in real time we have to subtract the red component from the
gray scale image to extract the red components in the image.
3 Use a median filter to filter out noise.
4 Convert the resulting gray scale image into a binary image.
5 Display the image.
6 Again a loop is used to bound the red objects in a rectangular box.

Step5: Stop the video acquisition.
Step6: Flush all the image data stored in the memory buffer.
Step7 Clear all the variables.

Fig. 8. Flowchart diagram for algorithm

VI. CONCLUSION

The conclusion of this paper proposes the idea of using Front Camera of a mobile phone to detect a color (Red, as presented above), convert the color into an array, process it as a signal then track its position at different instances in a video and generate output of its positions as an image file.

VII. REFERENCES

ABSTRACT:- Now days, the huge amount of data has been stored in educational databases increasing rapidly. The educational databases contain hidden useful information with many important factors related to the student’s learning. Data mining techniques have been applied to analyze this data and bring out the hidden knowledge. In this paper we have been shown a performance evaluation of different data mining classification techniques through different data mining tools through educational dataset of B.Tech students collected from a private Engineering College. The different data mining classification techniques would be applied and tested on the educational data sets. The parameters have been used percentage of accuracy and error rate of every applied classification technique. The proposed methodology would be classifying the data sets more accurately and efficiently.

Keywords—Data Mining, Classification, Tanagra, Orange, WEKA

1. INTRODUCTION

Now days the data mining is a relatively young and interdisciplinary area of computer science. It is a technique that attempts to discover new patterns in huge datasets. Different types of mining algorithms have been proposed by different researchers in recent years. A single algorithm may not be applied to all applications due to difficulty for suitable data types of the algorithm. Therefore the selection of a correct data mining algorithm depends on not only the goal of an application, but also on the compatibility of the dataset. In this paper we have been tested on the different data mining algorithms with the different tools by using the educational large dataset taken from private Higher Education Institutes.

1.1 DATA MINING

Data mining is a technique that is used to extract knowledge from huge amount of data. Data warehousing is the storage of data from different sources in the organization and data mining is the exploration of data from the data stored in the data warehouse. Data mining is the process of analyzing data from different views and converts this data into useful and meaningful information. The extract information can be used to increase the revenue and cut the cost [1].

Due to the importance of extracting information from the large data sets, data mining has become an essential component in various fields of human life [2]. In the current scenario a huge data is available in the industry and in educational organization. This large amount of data needs to convert into useful information and knowledge. Data mining is a technique which converts this raw data into information. Data mining is also call “Knowledge mining from data”, “Knowledge extraction”, “Data/pattern analysis”, “Data dredging” and “Data archaeology”[3].

Data mining consists of three steps which are shown in figure 1.

(i) Capturing and storing the data
(ii) Converting the raw data into information
(iii) Converting the information into knowledge

Fig. 1: Data Mining Process

1.2 ARCHITECTURE OF A DATA MINING SYSTEM

The architecture of data mining system is shown in figure 2 below.
In the above figure 2, data warehouse, world wide web and other information repositories contain set of database or information. After that data cleaning and integration techniques are applied on the give data sets. Knowledge base is used to search the interesting pattern. Data mining engine contain the set of functional modules for characterization, association, correlation analysis, classification, prediction, cluster analysis, outlier analysis and evaluation analysis. Pattern evaluation used to find out the interesting pattern. It interacts with data mining modules. User interface provide communication between users & data mining system. It specifies a data mining query[4].

2. DATA MINING TOOLS
The three best open source data mining tools have been performed the classification methods. Which are presented as follows: Tanagra, Orange, Weka.

2.1 TANAGRA
TANAGRA is free DATA MINING software for academic and research purposes [5]. It proposes several data mining methods from exploratory data analysis, statistical learning, machine learning and databases area. It runs under almost Windows Systems, in any case it has been tested under Windows 98, 2000, XP, Vista and Windows 7.

2.2 ORANGE
It is a component-based data mining and machine learning software suite that features friendly yet powerful, fast and versatile visual programming front-end for explorative data analysis and visualization, and Python bindings and libraries for scripting. It contains complete set of components for data preprocessing, feature scoring and filtering, modeling, model evaluation, and exploration techniques. It is written in C++ and Python[6], and its graphical user interface is based on cross-platform Qt framework.

2.3 WEKA
Written in Java, Weka (Waikato Environment for Knowledge Analysis) is a well-known suite of machine learning software [7] that supports several typical data mining tasks, particularly data preprocessing, clustering, classification, regression, visualization, and feature selection. Its techniques are based on the hypothesis that the data is available as a single flat file or relation, where each data point is labeled by a fixed number of attributes. Weka provides access to SQL databases utilizing Java Database Connectivity and can process the result returned by a database query. Its seven user interface is the Explorer, but the same functionality can be accessed from the command line or through the component-based Knowledge Flow interface.

3. RELATED WORKS
Several works have carried out by many researchers in the field of classification data mining. In recent years some of the works have recorded and presented in this section.

In [8] Carlos Marquez Vera et. al apply data mining technique to predict school failure and drop out. White box classification method such as induction rule and decision tree is applied to the middle school students. The accuracy of measuring student performance is improve by using all the available attribute, next selecting the best attribute and finally rebalancing data, and using cost sensitive classification.

In [9] Rahat Iqbal et. al focus on effective teaching and learning of educators. This can be increase by monitoring activity and performance. A fuzzy linguistic summarization technique is used for extracting linguistically interpretable scaled fuzzy weighted rules from student data.

In [10] Banumathi et al predicted the behavior of students from huge database is the major problem for education system. In this work the performance of students had analyzed through proposed UCAM clustering algorithm. UCAM algorithm removes the drawback of clustering algorithm. It set the threshold value of making unique cluster. This approach had reduced the overheads of fixing the cluster size and initial seeds as in K-means. This method improved the scalability and reduces the clustering error.

In [11] Monika Goyal et al was proposed a method to improve the efficiency of higher education institute by using data mining technique. Data mining techniques such as clustering, decision tree, and association rule mining had applied to higher education system to improve student performance.
In [12] Ryan S.J.D Baker has shown the major areas and trends in educational data mining. Educational data mining is an emerging discipline, concerned with developing methods for exploring the unique type of data that come from educational setting. It focused on application of educational data mining to web data, a perspective that accords with the history of the research area.

In [13] Richard A. Huebner focused on analyzing data to develop models for improving learning experience and improving institutional effectiveness. It provided some ways to apply analytical and data mining methods to educational related data. The emerging field of educational data mining examines the unique ways of applying data mining methods to solve educationally related problems. This work has focused on ways that data mining has used to improve student success and process directly related to student learning.

In [14] Chady El Moucary focuses on increase the academic performance of engineering student. The grades of these students are become important when they want to take admission in master course. A predictive model has also presented with a powerful decision making tool for these highly important issues. In this paper the author presented a study that aims at offering reliable and predictive tool for academic and administration working in engineering schools and universities to monitor student performance at an early stage of their educational path.

In [15] Feng Shi et al proposed the use of data mining technique such as association rule in college curriculum to help the teachers to arrange courses, scientific guidance for teaching and learning, improve the teaching management. In this article association rule of data mining was used in the university curriculum management.

In [16] C.P. Samaranayake focuses on the failure of a large number of talented students in the physical science stream of education advance level examination. The failure rate in the physical science stream is the highest among the entire four streams. This study determines the factor of high failure rate in physical science stream students of G.C.E. ‘A’ level examination by using data mining technique.

In [17] Sunita B Aher et al made a survey an application of data mining in education system and also present result analysis using WEKA tool. Different types of data mining task within the educational data mining such as classification, clustering, outlier detection, association rule and prediction. In this paper the performance of final year UG information technology course student is analyze and present the result which is achieved using WEKA tool.

In [18] Manpreet Singh Bhullar represents the problem faced by higher education institute. This higher education institute faced many challenges today like predicting the path of students, which student will enroll in which course, which student will need more assistance in particular subject. Data Mining helps the institute to take more accurate decisions. Data mining is a better tool to predict the result of student. Data has been classified using WEKA tool and use J48 algorithm to predict the result of the student. J48 algorithm offers a superior stability between precision, speed and interpretability of results. J48 classifies the data in the form of decision tree. From this decision tree we are easily identified the weak students.

In [19] M. Sukanya et. al had used classification and clustering algorithms of data mining for the performance improvement in education sector. By using these algorithms an educational institute could predict the number of enrolled students. Bayesian Classification methods have been use on student database to predict the student division on the basis of previous year database.

In [20] Shelly Gupta et. al, presented a study of different data mining classification techniques implemented on four different healthcare datasets, wherever the experiments were conducted by using the 10 fold cross validation method.

In [21] Pandya Sohi D. Et al utilized the untouched or hidden data of university to improve the performance of student. Some data mining techniques were used to analysis this hidden data. This work compared the classification and association rule mining technique to uncover the hidden pattern in an Indian university. The experiment results shows that classification comparatively bought more accurate and clear picture of hidden patterns.

5. METHODOLOGY USED

1. In this paper we have been carried out the performance evaluation of various data mining classification techniques on the basis of student’s dataset.
2. This work will be helpful to determine the classification techniques in terms of accuracy and error rate in vis-a-vis to the participating techniques on educational data sets.

3. The classification techniques which are to be tested such as: KNN, Naive Bayes (NB), SVM, CART, MLP, ID3 and Decision Trees.

4. The training performance of these techniques or algorithms to be measured according to their accuracy.

5. Some of the popular and powerful data mining tools such as Weka, Tanagra, and Orange are used in this work to predict the performance of the techniques on educational dataset.

6. This work will help to the researchers to determine the better results from the available data within the datasets after knowing the most suitable classification techniques for educational dataset.

5.1 Experimental Evaluation

We have been collected the huge data of B.Tech students from a private engineering college of Bharatpur, Rajasthan. This work focuses on the data mining and major issues associated with it. The aim of this work is how data mining techniques implemented and tested on student dataset called educational dataset.

5.1.1 Dataset Description

We have been tested the three dataset files by using three data mining tools in this work. The detailed description of the three datasets is shown in Table-4, Table-5, Table-6. (Appendix-I to Appendix-III)

5.1.2 Data Mining Tools Used

Weka, Tanagra, Orange

5.1.3 Parameters Used for Data Mining Analysis

Two Accurate rate, Error rate

5.1.4 Results for Classification Techniques

In this section, we have applied the different data mining classification techniques through different three data mining tools on the educational datasets of different B.Tech streams and results are shown in the different three tables.

### Table-1: Results obtained by Weka

<table>
<thead>
<tr>
<th>Techniques used</th>
<th>Accurate rate</th>
<th>Error rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>J48</td>
<td>76.12</td>
<td>22.20</td>
</tr>
<tr>
<td>Logistic</td>
<td>75.20</td>
<td></td>
</tr>
<tr>
<td>BN</td>
<td>74.89</td>
<td></td>
</tr>
</tbody>
</table>

### Table-2: Results obtained by Tanagra

<table>
<thead>
<tr>
<th>Techniques used</th>
<th>Accurate rate</th>
<th>Error rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>J48</td>
<td>76.12</td>
<td>22.20</td>
</tr>
<tr>
<td>Logistic</td>
<td>75.22</td>
<td></td>
</tr>
<tr>
<td>BN</td>
<td>76.45</td>
<td></td>
</tr>
<tr>
<td>NB</td>
<td>74.51</td>
<td></td>
</tr>
<tr>
<td>MLP</td>
<td>69.66</td>
<td></td>
</tr>
</tbody>
</table>

### Table-3: Results obtained by Orange

<table>
<thead>
<tr>
<th>Techniques used</th>
<th>Accurate rate</th>
<th>Error rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>J48</td>
<td>76.12</td>
<td>22.20</td>
</tr>
<tr>
<td>Logistic</td>
<td>78.26</td>
<td></td>
</tr>
<tr>
<td>BN</td>
<td>75.12</td>
<td></td>
</tr>
<tr>
<td>NB</td>
<td>77.21</td>
<td></td>
</tr>
<tr>
<td>MLP</td>
<td>69.23</td>
<td></td>
</tr>
</tbody>
</table>

6. CONCLUSION

Classification is an important data mining technique with broad applications. Classification has been used in every field of our life. It classifies data of different kinds. In this paper, we have tested different classification techniques to classify the data into classes with the help of the training datasets. The Comparison of results of different data mining tools for best suitable classification is as follow:

<table>
<thead>
<tr>
<th>Tools</th>
<th>Techniques</th>
<th>Accurate Rate</th>
<th>Error Rate</th>
</tr>
</thead>
<tbody>
<tr>
<td>Weka</td>
<td>SVM</td>
<td>21.98</td>
<td>22.20</td>
</tr>
<tr>
<td>Tanagra</td>
<td>C4.5</td>
<td>20.87</td>
<td>22.18</td>
</tr>
<tr>
<td>Orange</td>
<td>SV</td>
<td>21.13</td>
<td>20.76</td>
</tr>
</tbody>
</table>

The experimental evaluation shows that different classification techniques behave differently on educational dataset collected from an engineering College.

In future, more work may also be done on different datasets of different educational institutions using different parameters and techniques.

REFERENCES

[1] Jiawei Han and Micheline Kamber, “Data mining Concepts and Techniques”, Published by Morgan Kaufmann, 2006.


String Algorithms for Counting DNA Nucleotides, Transcribing DNA to RNA and Complementing a Strand of DNA

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Abstract - DNA (deoxyribonucleic acid) is found in all living organisms human, animals, bacteria and many viruses have only four nucleobases: adenine (A), cytosine (C), guanine (G), and thymine (T). RNA (ribose nucleic acid) is different from DNA because it contains a base called uracil in place of thymine. James Watson and Francis Crick proposed DNA structure as: DNA molecule made of two strands running in opposite directions, twisted together in a long spiral structure called a double helix and Adenine always bonds with thymine, and cytosine always bonds with guanine. This article focuses on string algorithms for Counting DNA nucleotides, Transcribing DNA to RNA and Complementing a strand of DNA.

Keywords— DNA String algorithms, Complementing a DNA Strand.

I. INTRODUCTION

Bioinformatics is a management information system for molecular biology. [1]. It is an Information Technology field that develops methods for storing, retrieving, organizing and analyzing biological data. Bioinformatics uses many areas of computer science, statistics, mathematics and engineering to process biological data. Complex machines are used to read in biological data at a much faster rate than before. Databases and information systems are used to store and organize biological data. Analyzing biological data may involve algorithms in artificial intelligence, soft computing, data mining, image processing, and simulation. The algorithms in turn depend on theoretical foundations such as discrete mathematics, control theory, system theory, information theory, and statistics. Commonly used software tools and technologies in the field include Java, C#, XML, Perl, C, C++, Python, R, SQL, CUDA, MATLAB, and spreadsheet applications. [4].

The aims of bioinformatics are three-fold. First, at its simplest bioinformatics organizes data in a way that allows researchers to access existing information and to submit new entries as they are produced, e.g. the Protein Data Bank for 3D macromolecular structures. The second aim is to develop tools and resources that aid in the analysis of data. For example, having sequenced a particular protein, it is of interest to compare it with previously characterized sequences. The third aim is to use these tools to analyze the data and interpret the results in a biologically meaningful manner. [3].

DNA is an acronym for the molecule deoxyribonucleic acid. DNA is contained in each living cell of an organism, and it is the carrier of that organism’s genetic code. The genetic code is a set of sequences which define what proteins to build within the organism. Since organisms must replicate and/or reproduce tissue for continued life, there must be some means of encoding the unique genetic code for the proteins used in making that tissue. The genetic code is information which will be needed for biological growth and reproductive inheritance. DNA is a double-stranded, helical molecule often called a “double helix”. Each strand is composed of a sequence of nucleotides. The nucleotide sequence is what encodes genetic information [2].

In the mid-20th Century, culminating in 1953 with a publication in Nature of fewer than 800 words by James Watson and Francis Crick proposed the following structure for DNA: The DNA molecule is made up of two strands, running in opposite directions. Each base bonds to a base in the opposite strand. Adenine always bonds with thymine, and cytosine always bonds with guanine; the complement of a base is the base to which it always bonds; see figure 1. [5]. The two strands are twisted together into a long spiral staircase structure called a double helix; see figure 2. [5]. DNA, in which the sugar is called deoxyribose, and the only four choices for nucleobases are molecules called adenine (A), cytosine (C), guanine (G), and thymine (T). Nucleotide is used as a unit of strand length (abbreviated to nt). [5]

II. PROBLEM: COUNTING DNA NUCLEOTIDES

A string is simply an ordered collection of symbols selected from some alphabet and formed into a word; the length of a string is the number of symbols that it contains. [5]

An example of a length 21 DNA string (whose alphabet contains the symbols 'A', 'C', 'G', and 'T') is "ATGCTTCAGAAAGGTCTTACG."
Given: A DNA string $s$ of length at most 1000 nt.
Return: Four integers (separated by spaces) counting the respective number of times that the symbols 'A', 'C', 'G', and 'T' occur in $s$.

**Algorithm**

```csharp
string strInput = txtInput.Text;
string strError = "";
int A = 0, C = 0, G = 0, T = 0;
for (int i = 0; i < strInput.Length; i++)
{
    string strN = strInput.Substring(i, 1);
    if (strN == "A")
        A++;
    else if (strN == "C")
        C++;
    else if (strN == "G")
        G++;
    else if (strN == "T")
        T++;
    else
        strError += strN;
}
lblResult.Text = A.ToString() + " " + C.ToString() + " " + G.ToString() + " " + T.ToString();
```

**Sample Dataset**

```
CAAAATCTGACTATTACGACCATTCTCTCT
TAAAAAGAGATGATACATCATGACCACACA
CTGCGT
```

**Sample Output**

```
28 19 10 20
```

III. PROBLEM TRANSCIBING DNA INTO RNA

A second nucleic acid exists alongside DNA in the chromatin; this molecule, which possesses a different sugar called ribose, came to be known as ribose nucleic acid, or RNA. RNA differs further from DNA in that it contains a base called uracil in place of thymine; structural differences between DNA and RNA are shown in Figure 3.[5]

The primary structure of DNA and RNA is so similar because the former serves as a blueprint for the creation of a special kind of RNA molecule called messenger RNA, or mRNA. mRNA is created during RNA transcription, during which a strand of DNA is used as a template for constructing a strand of RNA by copying nucleotides one at a time, where uracil is used in place of thymine. [5]

In eukaryotes, DNA remains in the nucleus, while RNA can enter the far reaches of the cell to carry out DNA's instructions.
An RNA string is a string formed from the alphabet containing 'A', 'C', 'G', and 'U'.

Given a DNA string t corresponding to a coding strand, its transcribed RNA string u is formed by replacing all occurrences of 'T' in t with 'U' in u.

**Given:** A DNA string t having length at most 1000 nt.

**Return:** The transcribed RNA string of t.

**Algorithm**

```csharp
string strInput = txtInput.Text;
strInput = strInput.Replace('T', 'U');
lblResult.Text = strInput;
```

**Sample Dataset**

```
TTAATACGGCTCGTCGACGAATGCGGTGACCGCCA
CGATAGTGGGGAAAGCGTCCTACTATCCTGGGTAC
ATTGT
```

**Sample Output**

```
UUAAUACGGCUCGACGAAUGCGGUG
ACCGCCAGAAUGGGGAAAGCGGCUCU
ACUAUCGGUACUCAGAC
```

---

IV. PROB. COMPLEMENTING A STRAND OF DNA

Watson and Crick's model, the bonding of two complementary bases is called a base pair (bp). Therefore, the length of a DNA molecule will commonly be given in bp instead of nt. By complementarily, once we know the order of bases on one strand, we can immediately deduce the sequence of bases in the complementary strand. These bases will run in the opposite order to match the fact that the two strands of DNA run in opposite directions. [5]

In DNA strings, symbols 'A' and 'T' are complements of each other, as are 'C' and 'G'.

The reverse complement of a DNA string s is the string sc formed by reversing the symbols of s, then taking the complement of each symbol (e.g., the reverse complement of "GTCA" is "TGAC").

**Given:** A DNA string s of length at most 1000 bp.

**Return:** The reverse complements sc of s.

**Algorithm**

```csharp
string strInput = txtInput.Text;
string strReverse = ""
string strError = ""
for (int i = strInput.Length - 1; i >= 0; i--)
{
    string strN = strInput.Substring(i, 1);
    if (strN == "A")
        strN = "T";
    else if (strN == "T")
        strN = "A";
    else
        strError += strN;
    strReverse += strN;
}
lblResult.Text = strReverse + "    " + strError;
```

**Sample Dataset**

```
GCCGACCTTCGTTAATAGGGCAGCTTACGTGTCGACTGGGTGTAAGACCTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGTGACGTGACGCCAGGACTGGTGCGCTGCTGCGCTGCTGGTGGTGGT
```

**Sample Output**

```
TACGAACATGGGTTCCTACCTCAAGTAAGTGCTGAC
GCCAAGGTAATCGGGGATACTTTCCTAAAC
TTATGCTACCTTTGCGGAGGACGACGCTCTCCC
GAACCGGGCTTGTGGCTTGGGCTGGTGACGATTT
GAGGGTAGCTAGCTACGTTGAGTTGTGACTCTGT
TTAGATGTAAGTGTCTATGCTACCTGAC
```

---

V. CONCLUSION AND FUTURE SCOPE

These algorithms use only basic string operations. Readers/programmers/researchers could enhance the performance of the algorithms. Various methods have been used by researchers to solve these problems using string operations. But still we could design and develop new algorithms in order to achieve lower space complexity and time complexity. Our future work will be to solving more complex DNA problems using dynamic programming and provide a web interface to visualize the algorithm's work flow and solutions.

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Data Management
Leveraging artificial Intelligence benefits in control engineering

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ABSTRACT- current trends in control processes require strict quality control requirements for reduced risk and economical operation of the process. Applying artificial intelligence techniques to control process provides more effective tool for control engineer. The applications of AI in control engineering can be applied at every stage of process design and implementation of control algorithms. In this paper we will study various AI tools such as fuzzy logic, neural net, genetic algorithm, swarm intelligence and their applications that can be applied to process control engineering.

Key word: Fuzzy, GA, Neural

I Introduction

What does Artificial Intelligence (AI) mean Artificial intelligence (AI) is an emerging area of computer science and has become an essential part of the technology industry. It focuses on the creation of intelligent machines that can work and react like humans. That is, they are programmed to obtain certain traits of humans such as knowledge acquisition, reasoning, problem solving, learning new things, planning. The advantages of creating human like intelligent machines are-

- Can do even complex work that humans may struggle/can not do.
- Can complete task faster than a human can most likely
- can discover unexplored things. i.e. outer space
- Less errors and defects are there in the work of intelligent machines
- Function is infinite, work of intelligent machines is not limited to specific areas.

Applications of AI- Use of AI is not limited to certain specific areas but they can be broadly classified as-

1) Heavy industries and space- manufacturing processes are now being automated, controlled and maintained by computer system. This system can do hazardous and dangerous tasks also without any problem which otherwise would not be possible with human beings.
2) Finance- intelligent software are used by banks and investment companies nowadays to analyze the data and predict the trends in investment market and thus helps in decision making process.
3) Computer science- AI has created space for complex programming and games that can lead to ultimate path for users.
4) Aviation- expert systems are being used in aircrafts to watch the atmospheric pressure and system status.
5) Weather forecast- neural nets, branch of Ai is used to learn the trends of weather and thus predict the weather patterns.
6) Swarm intelligence- this studies how intelligence occurs in natural systems, such as studying relationships in nature.

AI development and applications in process control:

To understand this we should understand the concept of control and use of control in automation so that we can develop intelligent control systems. Control theory is an interdisciplinary branch of engineering that deals with the behavior of dynamic systems with inputs. There is external input to the system known as reference. external input of a system is called the reference. The controller manipulates the inputs of the system when one or more output variables of a system follows a certain reference over time. The controller then manipulates the inputs to a system to obtain the desired effect on the output of the system. The objective is to take the proper corrective actions from the controller which results in system stability.

Process control is an engineering discipline that deals with architectures, mechanisms and algorithms for maintaining the output of a specific process within a desired range. Control is extensively being used in industries which enables mass production of continuous processes such as oil refining, paper manufacturing, chemicals, power plants and many other industries. Process control enables automation, with which a small staff of operating personnel can operate a complex process from a central control room. Process control can either be open loop or can use feedback.

Open loop controller also called non feedback controller computes its input into a system using only the current state and its model of the system. It does not use feedback to determine if its output has achieved the desired goal of the input.

Closed loop controller also called feedback controller utilizes feedback to obtain a more accurate or more adaptive control. So the output of the system is fed back to the inputs of the controller to make decisions about changes to the control signal that drives the plant.

The control theory and process control has given birth to the Automation or automatic control, which involves the use of various control systems for operating equipments such as machinery, processes in factories, boilers and heat treating ovens, switching in telephone networks, steering and stabilization of ships, aircraft and other applications. The biggest benefit of automation is that it saves labor, energy, reduces human intervention to improve quality accuracy and precision.
Web-enabled Performance Management Process

The web-enabled performance management process is a system that allows employees of the company to complete assessments and appraisals online and consolidated reports can then be automatically generated and sent to key stakeholders in the business.

Automated Insurance Quote System

An automated customer quote engine is used by insurance company which takes a series of parameters and uses pre-defined internal logic to generate a complex quote document for the client.

Integrated RFID stock tracking/fulfillment system

An integrated stock tracking and fulfillment system using RFID tags and linked to a database is used to track the presence and movement of stock in a warehouse. Intelligent control systems (control and intelligent systems)

Computers nowadays are being used in controlling and monitoring the processes in industries also. There is increased demand for managing and processing the data and control strategies for wider range of industrial applications. To achieve this, primary function of controllers can be improved by introducing AI, such as building expert system where the problem is considered sufficiently complex the expert system can think, analyze and reason much more than a human can do. Similarly fuzzy systems can be developed to deal with imperfect, imprecise knowledge of control systems.

The concepts of intelligence and control are closely related to each other. An intelligent system must define its goals and the methods to reach these goals. Control is required so that the system can move and achieve the desired goals in controlled manner and whenever it notices any deviation, it takes the corrective action to make intelligent system return back to the correct state. The term “intelligent control system” simply stresses the control aspect of the intelligent system.

There are three phases to develop intelligent control systems-

Phase 1- offline system- the development, design, coding and evaluation of system is done in offline mode. Such systems require inputs from keyboard and are used for training purposes. Their main function is to provide decision support to human operators based on process operation information.

Phase 2- online supervisory system- its objective is to evaluate intelligent systems interface with human operators, hardware instrumentation. The intelligent system is physically connected to the actual process and reports and suggestions are generated.

Phase 3- online closed loop system- this system directly reads the inputs from the process and returns the output back to the process. Objective of this is to reduce the operators routines tasks and improve safety and efficiency.

II. Methodology

AI technologies used for control engineering-Al consists of various technologies—expert systems, fuzzy logic, artificial neural networks, and genetic algorithms, among others. AI methods contribute to the high performance of complex, nonlinear processes and automation systems that would not be possible with traditional algorithms or equation-based controls.

Artificial neural net- neural network can be defined as: "a computing system made up of a number of simple, highly interconnected processing elements, which process information by their dynamic state response to external inputs." They are typically organized in layers, which consists of interconnected nodes and activation function. Inputs are presented to the network via the input layer, which in turn communicates to one or more hidden layers which does the actual processing. The hidden layers then passes the computations to the output layers where the answer is output.

Most ANNs work on the basis of learning rule which modifies the weights of the connections according to the input patterns that it is presented with. That is, ANNs learn by example as do their biological counterparts. The most common rule used by neural net is back propagation algorithm. Neural networks work differently for solving a problem as compared to conventional computers. Conventional computers use algorithms, which is set of instructions to solve a problem. Whereas, Neural networks process information in a similar way the human brain does. They learn and solve problems by example.

Neural net in matlab-

Workflow for Neural Network Design-
The work flow for the neural network design process has seven primary steps:

1 Collect the data
2 Create the network
3 Configure the network
4 Initialize the weights and biases
5 Train the network
6 Validate the network
7 Use the network

There can be any number of internal layers. Each layer takes some variable in form of vector u and transforms it into v, another variable. V can be achieved by multiplying Weight matrix w with u and adding some bias b.

\[ v = \text{sum}(w * u) + b \]

Creating a simple Neural FF Network in matlab

matlab inbuilt function newff will be used for generation of model.

First make a matrix R which is of 3 *2 size. First column will show the minimum of all three inputs and second will show the maximum of three inputs. In this case all three inputs are from 0 to 1 range.

SoR=[0 1; 0 1; 0 1];

Now size matrix is made having v-size of all layers-

\[ S = [5 1] \quad \text{(Take input layer v-size as 5)} \]

Now call newff func as-

\[ \text{Net} = \text{newff}([ 0 1; 0 1; 0 1], s, \{ \text{tansig}, \text{purelin} \}); \quad \text{(net is neural model, tansig, purelin shows mapping function of two layers)} \]

Now neural net has t be trained-

\[ \text{Net}=\text{train}(\text{net},I,O); \quad \text{(now net is trained)} \]

You can see the performance curve, as it gets trained.
So now simulate our neural network again on the same data and compare the outputs.

\[ O1 = \text{sim}(\text{nt}, I); \]

\[ \text{Plot}(1:1000, 0, 1:1000, 01); \]

We can observe how closely the two data green and blue follow each other.

Neural nets in control engineering-

- Neural Networks are used for cross validation, where a soft sensor is required to cross-validate the performance of a physical sensor or when the output must be determined by a series of lab analyses.

- Neural nets are used to monitor the process system with non linear characteristics. Data fro sensors are used as input for the network and the output such as temperature, pressure need to be monitored. Faults in process are monitored with neural net and corrective action is then taken.

- Neural net are used for process control, and is trained to estimate the unknown non linear process. This is generally used in identifying product composition correctly during operation of plant.

- Neural nets are used for fault detection for optimal operation of process plant. These systems use sensor measurements and process alarms as inputs and fault categories represent the output.

Fuzzy logic in control engineering-

- Fuzzy logic can be blended with conventional control techniques and adaptive techniques such as ANFIS can be developed for certain applications.

- Fuzzy logic are generally used in commercial appliances and their control. These systems use fuzzy logic thermostats to control the heating and cooling, this saves energy by making the system more efficient.

- Other significant application of fuzzy logic is in industrial automation. Fuzzy logic based PLC’s (process logic controller) can be used to control any number of industrial processes.
- Fuzzy logic is also used in decision making process such as signal processing or data analysis. An example of this is a fuzzy logic system that analyzes a power system and diagnoses any harmonic disturbance issues.

Genetic algorithm- Genetic Algorithms are adaptive, heuristic search algorithms based on idea of natural selection and natural genetics. They are based on the survival of fittest among structures to form a search algorithm. The algorithm maintains a set of initial population from which chromosomes based on fitness values are selected. The chromosomes with better fitness values are selected and crossover is done to produce new offspring. Offspring inherit characteristics from each parent. After crossover, mutation is done to improve the fitness of the individuals by randomly changing the value of a variable. Individuals in the population die and are replaced by the new solutions creating a new population. GAs are useful and efficient when the search space is large, complex or poorly understood. This process of crossover and mutation will continue till the optimum solution is found.

Outline of the Algorithm
The following describes how the genetic algorithm works:

1. The algorithm begins by creating a random initial population.

2. Sequence of new population is created. Use the individuals in the current generation to create the next population. To create the new population, the algorithm performs the following steps:
   - Compute fitness value of current population.
   - Selects members having higher fitness function value, called parents, based on their fitness.
   - Individuals in the current population that have lower fitness are chosen as elite. These elite individuals are passed to the next population.
   - Produce children from the parents. Children are produced either by making random changes to a single parent—mutation—or by combining the vector entries of a pair of parents—crossover.
   - Replace the current population with the children formed for the next generation.

3. The algorithm stops when optimized solution is formed.

Genetic algorithm in control engineering-

- These are widely used in off-line design applications such as controller design, model identification, system reliability, fault diagnosis.
- Other use of evolutionary algorithms is in tuning of PID controllers. These are used to control the working of any system and take corrective actions whenever error is diagnosed.
- They are used to optimize the existing controller parameters so as to replace the tried and tested existing methods.

- The evolutionary algorithms have been developed to optimize various aspects of intelligent systems. EA can be used to generate rule base in fuzzy systems and can function as alternative choice to learning weight in neural net.
- They have been used to improve reliability of the systems. They have also been designed to search for optimal design configurations using fault tree approach.

Hybrid systems- Intelligent systems include neural networks (NN), fuzzy systems (FS) and genetic algorithms (GA). Each of these systems possesses certain properties such as ability to learn, modeling, classifying, obtaining empirical rules, fitting specific kind of applications. Combination of these intelligent systems can be done to create neuro-fuzzy system, fuzzy-GA system, neuro-GA system and these systems together are called hybrid intelligent systems (HIS). Since every intelligent system have their own benefits and uses which make them beneficial for some particular applications and not all. So, they are combined to get the benefit of each tool together. The hybrid systems have found their applications in many areas such as process control, engineering design, financial trading, medical diagnosis.

Neuro-fuzzy system is realized as a neural network, in which fuzzy system parameters are encoded in several layers. This system uses fuzzy sets combined with neural network combining benefits of both. Neuro fuzzy system can handle any kind of information, imprecise data, has self learning, self organizing, self tuning capability. Fuzzy-GA system provides fuzzy system parameters optimization using GA. Optimization abilities of GA are used to develop the rules used by fuzzy sets. Neuro-GA system provides neural network parameters optimization using GA.

Swarm intelligence- it is a computational intelligence technique and it depicts the behavior of decentralized, self organized systems. They are used to solve real life optimisation problems that originally took its inspiration from the biological examples by swarming, flocking. The characterizing property of a swarm intelligence system is its ability to act in a coordinated way without the presence of a coordinator or of an external controller. These systems compose of parallel action that can perform different actions in different places at the same time. Another important property of swarm intelligence is fault tolerance. Fault creating part can be substituted by another one that is fully functioning.

III CONCLUSION
Intelligent control is the discipline in which control methods are developed in an attempt to emulate the important characteristics of human intelligence such as adaptation, planning, learning, dealing with uncertainty. Intelligent control methodologies are being applied to different areas such as robotics and automation, communications, manufacturing, traffic control and many others. Neural networks, fuzzy control, genetic algorithms, planning systems, expert systems, hybrid systems are the certain areas which can be applied to control engineering to develop the intelligent systems. The artificial intelligence field provides knowledge representation ideas, methodologies and tools such as semantic networks, frames, reasoning techniques to develop intelligent systems and control engineering help intelligent systems to learn and adapt, give optimized results which

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increases the efficiency of the system and its reliability thus increasing the areas of application of the system.

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Multicast Model based on grouping for Access Control in Online Social Networks

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Abstract— Social networking applications have become a very popular for communication and interaction, and participation of user has growing tremendously. Currently online social networks provide simple access control mechanisms with selected users to govern only access to information contained in their own spaces. Social networking sites allow users to restrict access to shared data, users, unfortunately, there is no any mechanism to providing the privacy concern over data associated with a group. In this paper, we have proposed a multicasting model for securing the data on group, And we tried to deal with the more comprehensive privacy conflict resolution with the help of this model. This model is used for collaborative management of shared data on online social networking sites. Besides this, we present a logical representation of multiparty access control model to perform the various features of existing logic and also perform various analysis tasks on our model.

Keywords— Social network, multiparty access control, access control & security, multicast model and policy specification.

I. INTRODUCTION

In recent year, we have seen unprecedented growth in Online Social Networking sites. We all know that facebook is one of the representative social networking sites claim that it has over 300 million users. To protect user data, access control provide central features or OSN.

OSN’s such as facebook and google+ etc. are designed to people for share the personal and public information and make social connection with friends, family, and strangers and so on. OSN means of interactions among people in which they create, share, and/or exchange information and ideas in virtual communities and networks. OSN is used to create highly interactive platforms through which individuals and communities share, co-create, discuss, and modify user-generated content [2][3][11].

II. USE OF OSN’S

Social Network has the ability to stay in touch with friends and family members from anywhere in the world has millions of people caught up in the excitement of social networking. Because social networks are where the customers are, many enterprises are also turning to social networks as a free and powerful means of communication.

Although OSNs provide access control mechanism for users to govern access to only their information contained in their own space, unfortunately users have no control over data residing outside their space. While each user contains profile information and information may be public or private. So, users don’t want to share their information with public.

OSNs provide some space to each user for basic profile and sharing photos and videos with others. When a users share their content and photo on a group, every member of group are able see and share that. So now these users want more privacy because they don’t want to share their information

III. BACKGROUND & RELATED WORK

In this section we provide a short introduction to work in the area of mobile social networking and the technologies that have made it possible.

• Social Networks

A social networking service is an online service, platform, or site that focuses on facilitating the building of social networks or social relations among people who, for example, share interests, activities, backgrounds, or real-life connections. A social network service consists of a representation of each user (often a profile), his/her social links, and a variety of additional services. Most social network services are web-based and provide means for users to interact over the Internet, such as e-mail and instant messaging. Online community services are sometimes considered as a social network service, though in a broader sense, social network service usually means an individual-centered service whereas online community services are group-centered. Social networking sites allow users to share ideas, activities, events, and interests within their individual networks.

Social networking is the grouping of individuals into specific groups, like small rural communities or a neighborhood subdivision, if you will. Although social networking is possible in person, especially in the workplace, universities, and high schools, it is most popular online. This is because unlike most high schools, colleges, or workplaces, the internet is filled with millions of individuals who are looking to meet other people, to gather and share first-hand information and experiences about cooking, golfing, gardening, developing friendships professional alliances, finding employment, business-to-business marketing and even groups sharing information about baking cookies to the Thrive Movement.

In [1] has discussed about the possible misuses of online social network. And also presented techniques for building malicious applications in social networks that can launch DDoS attacks using a social network. The main goal was to highlight possible miss-uses of current technologies deployed in social networks. Social networking websites has to provide space on their own servers to application developers to develop their applications so that they can scams whenever required.

In [2] OSNs often use user relationship and group membership to distinguish between trusted and untrusted
users. For example, in Facebook, users can allow friends, friends of friends (FOF), groups, or public to access their data, depending on their personal authorization and privacy requirements. Users unfortunately have no control over data residing outside their spaces. When a user uploads a photo and tags friends who appear in the photo, the tagged friends cannot restrict who can see this photo, even though the tagged friends may have different privacy concerns about the photo. Stakeholders (tagged friends) don’t want to share their information with public. In this paper, they pursue a systematic solution to facilitate collaborative management of shared data in OSNs. They produce a Multiparty Access Control (MPAC) for sharing the data on OSNs can undermine the protection of user data. Their (MPAC) model contains a multiparty policy specification scheme. The use of an MPAC mechanism can greatly enhance the flexibility for regulating data sharing in OSNs.

In [3] has introduced that user participation on online social network has increased tremendously. The types of data uploaded and shared on user profiles also include sensitive information. And highlights the potential attacks owing to the vast amount of user personal information available on social networks. And proposed a theoretical model for resolving the problems associated with the current default privacy and wider accessibility design implemented by most social networks. This paper has also discussed the prevailing attacks on social network users.

In [4] Social networking websites are using tremendously, many people are not properly aware of the risk with using these websites and applications and examines the issues of security, privacy and trust in online social networking sites with using viewpoints. In this, they have considered two countries like Thailand and UAE both countries have witnessed of using social networking sites. They have three instruments like survey question interviews are use to better understanding the result. After survey of two countries, they said especially women are felt more comfortable using social networking sites.

In [5] Social networking is used in and outside every organization. There are many social networking whites as sites as facebook, twitter, orkut etc and issues in different way as chatting, messaging, games, video, photo upload etc. however, everybody observed that these are many user face different problems as identity theft stealing of personal information. Authors have discussed on various kinds of security issues authors main focused is on the many issues of security and also dual with possible solutions on issues. User said that almost 25.61% users of social networking wed sites are number aware of the security issues.

In [6] It was a small survey about the online social network. In this survey, authors have focused on the privacy aspect and their concerned on the possible attacks. Because users share their data on social networks without bringing aware of consequences. Every users profile contains the sensitive information and users as advertisement. So attackers can take the advantage of it. Authors have discussed a preserving privacy in social network data and identity a

privacy attacks as neighbourhood attacks by mathematical formulation and computational models for security and privacy. In [7] have proposed a new security architecture called socially keyed (sokey) architecture achieving zero possibility for personal information leak from Social networking sites. This architecture is very confidential to make sure that providing information on social networking sites will never leak.

In [10] Social networking is the easiest way for communication. Social networks contains the millions of user each user have their own profile that contain more information. Users share so much data on social networking sites and this action became the target of attacks. Attackers found the very easy way to steal the information through these networking sites. In this authors have proposed a architecture for securing the information between user and a secure request response architecture.

In [11] have proposed a new trust framework for social network. Social networks carry important information and these are become the target of various attacks. In this paper, authors have discussed two attack models simple and intelligent. They evaluate the filtering effect on trust framework. Attacker may be terrorist, friends and etc. where attackers send the malicious information. Authors have discussed about the advertising company and want to promote their product, for promotion of product company give the offer to limited chairs and than those users give their positive feedback.
In [12] has discussed about the real life security issues and threats with facebook. They have discussed different type of facebook security issues as privacy and confidentially, authentication and identity theft, intellectual property theft, vandalism, harassment & stalking, data motion & disparagement, spam and cyber squatting. There all risks are greatest issues for facebook because fact is that people trust their facebook friends means that identity theft is greater.

In [13] have proposed a privAware tool to detect unintended information loss in online social networks to identify privacy risk and provide solution to reduce information loss because measuring the privacy risk in online social networks is a big challenge. Millions of users are participating in social networking sites and share the data in a huge large amount.

### IV. ACCESS CONTROL & SECURITY

**Multiparty Access Control (MPAC) for OSNs:**

Multiparty access control (MPAC) can undermine the protection of user data. In this we analyze three scenarios like profile sharing, content sharing and relationship sharing to understand the problems posted by the lack of security control in OSNs.

**Profile Sharing.** A user accesses the profile attributes of his/her user’s friend. In this situation both the user and his/her friend want to gain control over the profile attributes. In Fig.1a A disseminator can share owner’s profile attributes to an accessor [2].

**Relationship Sharing.** In Fig.1b Owner has a relationship with another user called stakeholder. Stakeholder shares the relationship with an accessor [2].

**Content Sharing.** In Fig.2a Owner of a content shares the content with other members of OSN, and the content has multiple stakeholders who may also want to involve in the content of content sharing. In Fig.2b Contributor publishes a content to other’s space and the content may also have multiple stakeholders. In Fig.2c Owner or contributor publishing the content, and then a disseminator can view and shares the content [2].

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**MPAC Policy Specification**

It is very essential for MPAC policies to regulate access and representing authorization requirements from multiple associated users to enable a collaborative authorization management of data sharing in OSNs.

**Accessor Specification:** Accessor is the set of users who granted to access the shared data. Accessor can be represented with a set of user names, relationship names and group names in OSNs.

The accessor specification is defined as a set, accessor = \{a1, a2, . . . , an\}.

**Data Specification:** The data specification represented in three ways; profile, relationship and content sharing. For effective privacy the different controllers provide sensitivity levels on data.

**MPAC Policy**

The multi party access control policy is a 5 - tuple \( P = \langle \text{controller}, \text{Ctype}, \text{accessor}, \text{data}, \text{effect} \rangle \) where

- **U:** Controller is a user who can regulate the access of data.
- **CT:** Ctype is the type of the controller.
- **AC:** Accessor is the set of users who granted to access the shared data.
• D: Data is represents a data specification.
• Permit, Deny: Effect is the authorization effect of policy.

MPAC evaluated in step by step. Initially an access request goes to under policy evaluation, which is done under four controllers. The four controllers provide their own privacy policies in the form of decision either permit or deny in step-1 process. After giving decisions by individual controllers, they are aggregated and make final decision by using decision voting schemes in step-2 process. The final decision making decides whether the access request is allowed or refused.

Essential reason leading to the privacy conflicts is that multiple controllers of shared data have different privacy concerns.

As a user share their data on a group and multiple users are added in that group, every member of that group are able to share of user’s data.

As we know that at the data uploading time on facebook, we use the privacy and provide the permission that whose friend may be view our data or not.

We can use the MPAC model (a technique) to put the privacy within the group and can design an algorithm in the group privacy concern so that the user from the joined group only can use and see the data. Joined person can’t share the group data outside the group.

So, we want that group members only see the data but not able to access.

V. CONCLUSIONS

We have discussed a solution for collaborative management of shared data in OSNs. We have proposed a multicast model for sharing the data. In which users share their data on group for multiple users. Our experiment will proved low users awareness securing user’s data, and group members will not permitted for share and download the data, they are only permitted for view and accessing. Users will not afraid of setting up their personal information such as address and other personal information.

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Abstract: Increasing speed and complexity of design gives a significant increase in power consumption in VLSI chips. Speed, power consumption and space are major issues in VLSI circuit. To meet these challenges there are certain design techniques which are used to reduce power. Optimization of power can be done by considering various components such as transistor sizing, voltage scaling, variable VDD, multiple threshold voltages, voltage islands, power gating, long channel transistor, stacking & parking states and logic styles etc. Power consumption in VLSI circuit is data dependent. In this paper different design methods are tested to optimize the power. It is found that algorithm based design reduces as is of component used and their function different optimization techniques can be used. For example in case of multiplier power consumption depends upon data. This is because switching contributes to more power consumption. This can be optimized by using various gate combinations. Gate switching can be reduced by using algorithms. For example in case of multiplier design method Booth algorithm as well as modified Booth algorithm can be used for efficient multiplication. In this paper various approaches are used for power consumption eg. transistor sizing, voltage scaling, variable VDD, multiple threshold voltages, voltage islands, long channel transistor, stacking & parking states, logic styles, Genetic algorithm, Booth multiplication etc.

This paper is divided into five sections. First section is of introduction. Second section is about preliminary studies. Third section includes different design methods and their testing. Fourth section is of comparison with discussion. Finally, fifth section is conclusion of the work reported here.

II. PRELIMINERIES

2.1 Power Dissipation

In case of VLSI circuits power dissipation is of two types i.e. static power dissipation and dynamic power dissipation. Static power dissipation is due to three major factors as leakage current, sub threshold current of transistors and tunneling effect of current. So optimization can be achieved by concentrating all these design issues. As we know that MOS transistor has two P-N junctions: one is between source and drain and other is between drain and source. When source substrate junction or drain substrate junction is reversed biased. The leakage current flows through reverse biased P-N junctions. Generally this leakage current is very small in magnitude so we ignore this current. Leakage current is temperature dependent. This is because leakage current is caused by thermally generated carriers. With increase in temperature, leakage current increases. According to a survey it is found that it becomes double for every 10°C rise in temperature. From this it is clear that temperature of a circuit must be controlled at any cost.

Gate switching activity that results reduction of power in multiplier circuit.

Key Words: - Transistor sizing, voltage scaling, variable VDD, multiple threshold voltages, voltage islands, long channel transistor, stacking & parking states, logic styles, Genetic algorithm, Booth multiplication.

I. Introduction

The Reduction in power dissipation is an important design issue in VLSI circuits. The design parameters have major effect on the overall performance of the system. On the bas

2.1.1 Sub threshold current of transistors

At VGS = VTH surface is said to be inverted. In this condition there is strong inversion and conduction takes place afterwards. But practically VGS < VTH. MOS transistor is under subthreshold or weak inversion conduction. This current is very low. But if the thickness of dielectric is very good dielectric. The leakage current through this good dielectric is very low. Hence (PD)static will also be more. The threshold voltage has to keep high. As (PD)static = \( P(t) = i_{leak}(t) \times V_{DD} \)

Here \( i_{leak}(t) \) is the leakage current or static current and \( V_{DD} \) is supply voltage. From this it is clear that if leakage current increases, then static power increases. Hence threshold voltage represents a tradeoff between performance and static power dissipation.

2.2 Tunneling Current Effect

As gate oxide material is SiO2 and SiO2 is very good dielectric. The leakage current through this good dielectric is very low. But if the thickness of dielectric is very less. Then electrons can tunnel through this dielectric. If the thickness of dielectric decreases then tunneling effect increases.

2.3 Dynamic Power Dissipation

Dynamic power dissipation occurs in charging and discharging of capacitive loads. Let \( C_L \) is the capacitive load. If this capacitive load is switched between ground and \( V_{DD} \) and switching frequency is taken as (f)switching then charging of load taken takes place for T (f)switching time and discharging also takes place for same time, T being the time interval. During charging current flows from \( V_{DD} \) to \( C_L \) and during discharging current flows from \( C_L \) to \( V_{DD} \).

Total charge transferred from positive supply rail (\( V_{DD} \)) to negative supply rail (ground) during charging and discharging cycle will be:

\[ Q = C_L \times V_{DD} \]
III Design methods for optimization

3.1 Power Optimization as Data dependent

Complexity of data contributes to switching activity in the circuit. With the use of efficient algorithm design component of circuit can be reduced. By the application of simulation to standard design and comparing it with optimization better design component can be found. Gate switching for all initial states and all inputs can be simulated to analyze power consumption. Data dependency is helpful in gate design complexity. Ordering of gate inputs affect both power and delay. Prasad [1] has described methods to optimize the power and/or delay of logic gates based on transistor reordering. So, considerable improvements in power and delay can be obtained by proper ordering of transistors. For instance, late arriving signals can be placed closer to the output to minimize gate propagation delay. Another approach to reduce power is to consider the size of gate, which has significant impact on circuit delay and power dissipation. By increasing the size of transistors in a given gate, delay of the gate can be decreased but in contrast, power dissipated in the gate and fabrication space increases. Therefore, an optimum balance can be achieved by sizing of transistors appropriately. A method is to compute the slack at each gate in the circuit, where the slack of a gate corresponds to how much the gate can be slowed down without affecting the critical delay of the circuit. Alternatively, in different sub – circuits, where slack is greater than zero are utilized and the size of the transistors is reduced until the slack becomes zero, or the transistors attain a minimum size.

3.2 Combinational Gate Level Design

In gate level design of circuit, different combination of logical gates may different combination of logical gates may produce same circuit output but different value of power consumption. Path balancing, factorization and don’t care optimization may be utilized to optimize power consumption. Path balancing can be achieved by avoiding delay at each input gate. Genetic Algorithm can be used to determine different combination of gates and power consumption can be formulated by devising Fitness Function. Coello [3] has proposed design of combinational logic circuit based on Genetic Algorithm. By defining chromosome development schemes of various combinations of logic circuit can be developed using cross over and mutation. This approach is more efficient than human designer as various constraint of design circuit can be devised subject to fitness function. Genetic Algorithm can reduce number of gates, which consequently reduce power consumption; as the work of Coello [2], shows on 2-bit adder and 2-bit multiplier with a particular ‘cardinality’ that 56% reduction in number of gates for the circuit can be achieved.

IV. Comparison and discussion
Number of Gates versus power saving in CMOS – based on ISCAS-89 benchmark circuits. [8] is as shown below in Figure 1.

![Figure 1. Gate v/s power graph](image1)

<table>
<thead>
<tr>
<th>Power Reduction in supply voltage</th>
<th>Speed Loss</th>
<th>Constraints/Specifications</th>
</tr>
</thead>
<tbody>
<tr>
<td>leakage power reductions up to 54%</td>
<td>Not reported</td>
<td>“Logic design to reduce the leakage power of CMOS circuits that use clock gating to reduce the dynamic power dissipation tested on ISCAS-89 benchmark circuits” [8]</td>
</tr>
<tr>
<td>0.13 V or 800 times</td>
<td>15 times</td>
<td>“0.5-μm gate length and static logic” [9]</td>
</tr>
<tr>
<td>“1.1 V supply and consumes less than 5 mW which is more than three orders of magnitude lower power compared to equivalent commercial solutions.” [10]</td>
<td>Not reported</td>
<td>--</td>
</tr>
</tbody>
</table>

Table 2. Power reduction versus Speed Loss

For instance, a modular approach as shown below in Figure 2 is proposed to adopt appropriate optimization technique considering different possibilities in multiplier design.

![Figure 2. Modular approach for multiplier design](image2)

**CONCLUSION**

It is found that data complexity and various combination of gate level digital circuit has considerable impact in power dissipation. Besides, this physical design can be optimized by using Genetic algorithm by analyzing the placement option. This is subjected to optimum space allocation. Similarly selection of Booth algorithm may reduce power consumption of data complexity. It is found that in multiplier circuit Modified Booth Algorithm reduces power consumption as compared to other methods of multiplication.

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Review: Cache-timing attacks on AES

Abstract: The paper analyzes cache based timing attacks on optimized codes for Advanced Encryption Standard (AES). Cache-timing attack is a type of side channel attack that takes advantage of the information leaked by encryption devices. In this paper it has been reviewed that the main threat to the security is cache-timing attacks and there are various countermeasures provided to solve this problem.

Index Terms: Advanced Encryption Standard (AES), Cache-timing attacks, side channels, mitigation of timing attacks.

INTRODUCTION

In cryptography, a timing attack is a side channel attack in which the attacker attempts to compromise a cryptosystem by analyzing the time taken to execute cryptographic algorithms. Every logical operation in a computer takes time to execute, and the time can differ based on the input; with precise measurements of the time for each operation, an attacker can work backwards to the input. Timing attacks are often overlooked in the design phase because they are so dependent on the implementation. AES is not the only cryptographic function vulnerable to this type of attack but we need to provide special treatment to the background of AES and x86 memory/cache prior to moving on to the attack.

In a side-channel attack, an attacker recovers secret information from his victim by observing or manipulating a shared resource. The most attractive channels for such attacks are shared hardware resources such as the data cache, and the most devastating attacks recover cryptographic keys. Even as the traditional scenario for side channel attacks multiuser timesharing on workstation system has fallen into decline, other attractive attack scenarios have arisen.

The new challenges to AES side-channel attacks are:
- The AES-NI instruction set, which moves AES data structures out of the cache;
- Multicore processors with per-core L1 and L2 caches;
- The complexity of modern software and the pressure that it places on caches;
- The increasingly sophisticated and poorly documented pre-fetcher units on modern processors; and
- The switch from virtually tagged to physically tagged caches.

The rest two of these can make AES cache attacks impossible; the last three increase the difficulty of AES cache attacks and make them inapplicable in some settings.

II. AES ALGORITHM

AES is a symmetric key cryptography. The AES standard comprises three block ciphers, i.e. AES-128, AES-192 and AES - 256. The encryption of AES is carried out in blocks with a fixed block size of 128 bits each. The AES cipher calculation is specified as a number of repetitions of transformation rounds that convert the input plaintext into the final output of ciphertext. Each round consists of several processing steps, including one that depends on the encryption key. A set of reverse rounds are applied to transform the ciphertext back into the original plaintext using the same encryption key. AES operates on a 4x4 column major order, columns are listed in sequence(think flattening a matrix by following the column down and then moving to the next column), matrix. Key size, as stated before, defines the number of repetitions (in cycles) of transformations from plaintext to ciphertext. They are defined as:
- 128bit key = 10 cycles.
- 192bit key = 12 cycles.
- 256bit key = 14 cycles.

Key expansion: Round keys are derived using a key schedule. The key schedule relies on a few operations:
- Rotate: Rotate 32bit word 8bits to the left causing the high 8bits to wrap around.
- R-Con: In Rijndael's finite field, exponentiate to 2 of a user-specified value. This equates to a look-up of some constants.
- S-box: A lookup table that is generated by determining the multiplicative inverse for a given number in Galios Field (2^8).

InitialRound.

AddRoundKey - Each byte of the state is bitwise xor'd with the round key. For each round, a subkey is derived from the main key and is combined as shown in Figure 1.1
Rounds:

i) SubBytes - Non-linear substitution that replaces each byte with another according to a lookup table. Each byte is replaced with a SubByte using an 8-bit substitution box, or S-box. The S-box is derived from the multiplicative inverse over GF(2^8) to provide non-linearity to the cipher as shown in Figure 1.2

ii) ShiftRows - Transposition step that shifts each row cyclically. The first row is left unchanged and the shifting pattern is the same for 128bit and 192bit key sizes.

Figure 1.3: ShiftRows on AES

MixColumns - Mixing step, that operates on columns only, combining four bytes in each column. Uses an invertible linear transformation. Together with ShiftRows, this provides specifically AES, may leak timing information during cache hits or misses. Then he put forward an attack named cache-based timing attack, which can reconstruct the key by observing the data flow of different levels of cache.

Storage and CPU caches work like a lookup tables. As we are only interested in CPU caches, we will leave storage caches as an exercise to the reader. A CPU cache is used to reduce the average time to seek/access memory; intuitively, if you were to think about the performance of modern day CPUs, you might compare an i7 to an i5 where the clocks are similar, but the i7 is more capable due to a larger cache. From that, memory accesses from cache memory are faster (lower latency) than those from main memory.

More recently, Mowery showed that the AES cache-timing attack is no longer feasible now, for several reasons. First, the AES-NI instruction set moves AES data structures out of the cache. Secondly, multicore processors with per-core L1 and L2 caches are now used.

Let’s look at what Bernstein’s word are from the software developer’s point of view:

- AES scrambles 128bit input n.
- Generate a key, say 128bit, k.
- Define a constant 2048bit (256bytes) table S.
- Define a constant 2048bit table S’.
- S and S’ are expanded into four 8192bit (1024bytes) tables: T0, T1, T2, T3.

AES uses two 128bit (16byte) auxiliary arrays, x and y. diffusion to the cipher.

- Array x is initialized to k.
- Array y is initialized to n⊕k (initial key round as above).
- Array x is chunked into 4 4-byte arrays: x0, x1, x2, x3.
- Compute Array e as = (S[x3[1]]⊕1, S[x3[2]], S[x3[3]], S[x3[0]])
- Array x now replaces x0, x1, x2, x3 with (e⊕x0, e⊕x1⊕x0⊕x1⊕x2⊕x0⊕x1⊕x2⊕x3).

So the attacker basically has to:

- Passively watch the time taken by the victim to handle many input n;
- Total AES times for each possible n[13], and;
- Observe overall AES time when n[13] is a maximum value.

IV. METHODS OF MITIGATING CACHE BASED ATTACKS

There are various methods presented for the mitigation purpose against this type of attack. These are reviewed in this paper and are as follows:

A. Software Mitigations

The main aim of Bernstein[1] is to prevent against this type of attack appears to be write constant time AES software, and he outlines different methods by which to accomplish this. The solutions that are presented appear to be rather complex difficult to reach in addition to causing degradation in performance. The architecture of the system appears to affect the way the solution is reached; meaning the processor, cache, and register specification, and arrangements could potentially result in multiple software implementations. A point was made that incorporating the cryptographic functions into the kernel would allow for more control over what occurs during cryptographic actions.

Osvik, Shamir, and Tromer (OST)[2] identified additional techniques which could mitigate against this attack. A few of these are:

- Avoiding memory access
- Alternate lookup tables
- Data-oblivious memory access pattern
- Cache pre-loading

B. Hardware Mitigations

Usually if there is a solution in software, there is also one in hardware. In this case, the research conducted on mitigating against side channel attacks by making modifications to hardware, is limited. One technique which was designed with cache timing attacks specifically in mind was by Page. The basic premise is a different cache architecture, which partitions the cache in such a manner that prevents the cache from being used as a shared resource. This would prevent information that should not be removed from the cache from being removed. It should be noted that although there is not a lot of information available on this subject, it is a growing area of research solutions in hardware to the leakage of...
information by systems during cryptographic functions.

It was decided to use the smallest table lookup, 256 byte S-box equivalent lookup table. The architecture they deployed this on required four lines of cache. Prior to each encryption phase the table was fetched, and it is periodically permuted. Performance tests were conducted with different variations of this process with timing results recorded. Speed degradation is apparent in the most secure of the variations.

From all of these methods, the one that appears to have the most merit and has been experimentally demonstrated is the one presented by BGNS. The manner in which they propose to thwart the attack is general in nature and not architecture dependent and does not appear to be as costly from a performance standpoint.

V. CONCLUSION

In this paper, it is concluded that the main threat to the security is side-channel attacks. These attacks were classified by Bernstein any many of the people worked on it and tried their method to solve this problem. Various countermeasures have been proposed over the years to provide protection against cache timing attacks.

VI. REFERENCES

Cloud computing a fundamental change in computer architecture

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Abstract — Cloud computing is the latest effort in delivering computing resources as a service. It represents a shift away from computing as a product that is purchased, to computing as a service that is delivered to consumers over the internet from large-scale data centres – or “clouds”. Cloud computing is gaining growing popularity in the IT industry, academia appeared to be lagging behind the rapid developments in this field. This paper presents the detail of cloud computing architecture and characteristics of cloud computing. This gives the detail of IaaS, PaaS, SaaS and essential characteristics in this field. The aim of this paper is to explain the characteristics of next generation computer and why IT sector (organisations of all sizes and types) is switching to cloud computing.

Keywords: Cloud Computing, IaaS, PaaS, SaaS.

I. INTRODUCTION

The increased degree of connectivity and the increasing amount of data has led many providers and in particular data centers to employ larger infrastructures with dynamic load and access balancing. By distributing and replicating data across servers on demand, resource utilization has been significantly improved. Similarly web server hosts replicate images of relevant customers who requested a certain degree of accessibility across multiple servers and route requests according to traffic load. However, it was only when Amazon published these internal resources and their management mechanisms for use by customers that the term “cloud” was publicly associated with such elastic. Cloud systems have focused on hosting applications and data on remote computers, employing in particular replication strategies to ensure availability and thus achieving load balancing scalability, infrastructure especially with “on demand” access to IT resources.

II. DEFINITION

There has been much discussion in industry as to what cloud computing actually means. The term cloud computing seems to originate from computer network diagrams that represent the internet as a cloud. Most of the major IT companies and market research firms such as IBM [8], Sun Microsystems [1][4], Gartner [9] and Forrester Research [10] have produced whitepapers that attempt to define the meaning of this term.

The United States National Institute of Standards and Technology (NIST) has developed a working definition that covers the commonly agreed aspects of cloud computing. The NIST working definition summarises cloud computing as:

a model for enabling convenient, on-demand network access to a shared pool of configurable computing resources (e.g., networks, servers, storage, applications, and services) that can be rapidly provisioned and released with minimal management effort or service provider interaction [11]

The NIST definition is one of the clearest and most comprehensive definitions of cloud computing and is widely referenced in US government documents and projects. This definition describes cloud computing as having five essential characteristics, three service models, and four deployment models.

The essential characteristics are:

- On-demand self-service: computing resources can be acquired and used at anytime without the need for human interaction with cloud service providers. Computing resources include processing power, storage, virtual machines etc.
- Broad network access: the previously mentioned resources can be accessed over a network using heterogeneous devices such as laptops or mobile phones.
- Resource pooling: cloud service providers pool their resources that are then shared by multiple users. This is referred to as multi-tenancy where for example a physical server may host several virtual machines belonging to different users.
- Rapid elasticity: a user can quickly acquire more resources from the cloud by scaling out. They can scale back in by releasing those resources once they are no longer required.
- Measured service: resource usage is metered using appropriate metrics such monitoring storage usage, CPU hours, bandwidth usage etc.

The above characteristics apply to all clouds but each cloud provides users with services at a different level of abstraction, which is referred to as a service model.
The three most common service models are (figure 1):

- **Software as a Service (SaaS):** this is where users simply make use of a web-browser to access software that others have developed and offer as a service over the web. At the SaaS level, users do not have control or access to the underlying infrastructure being used to host the software. Salesforce’s Customer Relationship Management software3 and Google Docs4 are popular examples that use the SaaS model of cloud computing.

- **Platform as a Service (PaaS):** this is where applications are developed using a set of programming languages and tools that are supported by the PaaS provider. PaaS provides users with a high level of abstraction that allows them to focus on developing their applications and not worry about the underlying infrastructure. Just like the SaaS model, users do not have control or access to the underlying infrastructure being used to host their applications at the PaaS level3. Google App Engine5 and Microsoft Azure6 are popular PaaS examples.

- **Infrastructure as a Service (IaaS):** this is where users acquire computing resources such as processing power, memory and storage from an IaaS provider and use the resources to deploy and run their applications. In contrast to the PaaS model, the IaaS model is a low level of abstraction that allows users to access the underlying infrastructure through the use of virtual machines. IaaS gives users more flexibility than PaaS as it allows the user to deploy any software stack on top of the operating system. However, flexibility comes with a cost and users are responsible for updating and patching the operating system at the IaaS level. Amazon Web Services’ EC2 and S37 are popular IaaS examples.

Erdogmus [12] described Software as a Service as the core concept behind cloud computing, suggesting that it does not matter whether the software being delivered is infrastructure, platform or application, "it’s all software in the end" [12]. Although this is true to some extent, it nevertheless helps to distinguish between the types of service being delivered as they have different abstraction levels. The service models described in the NIST definition are deployed in clouds, but there are different types of clouds depending on who owns and uses them. This is referred to as a cloud deployment model in the NIST definition and the four common models are:

- **Private cloud:** a cloud that is used exclusively by one organisation. The cloud may be operated by the organisation itself or a third party. The St Andrews Cloud Computing Co-laboratory8 and Concur Technologies [13] are example organisations that have private clouds.

- **Public cloud:** a cloud that can be used (for a fee) by the general public. Public clouds require significant investment and are usually owned by large corporations such as Microsoft, Google or Amazon. Community cloud: a cloud that is shared by several organizations and is usually setup for their specific requirements. The Open Cirrus cloud testbed could be regarded as a community cloud that aims to support research in cloud computing [14].

- **Hybrid cloud:** a cloud that is setup using a mixture of the above three deployment models. Each cloud in a hybrid cloud could be independently managed but applications and data would be allowed to move across the hybrid cloud. Hybrid clouds allow cloud bursting to take place, which is where a private cloud can burst-out to a public cloud when it requires more resources.

**Fig. 1: Layer Architecture of cloud computing**

**Fig. 2: Cloud computing deployment and service models**

Others such as Vaquero et al. [15] and Youseff et al. [16] concur with the NIST definition to a significant extent. For example, Vaquero et al. studied 22 definitions of cloud computing and proposed the following definition:
Clouds are a large pool of easily usable and accessible virtualized resources (such as hardware, development platforms and/or services). These resources can be dynamically re-configured to adjust to a variable load (scale), allowing also for optimum resource utilization. This pool of resources is typically exploited by a pay-per-use model in which guarantees are offered by the Infrastructure Provider by means of customized SLAs.

This definition includes three of the five characteristics of cloud computing described by NIST, namely resource pooling, rapid elasticity and measured service but fails to mention on-demand self-service and broad network access.

III. CHARACTERISTICS OF NEXT GENERATION ARCHITECTURE

Characteristics of what we believe to be next generation architectures that will support substantive changes in global enterprise constructs and operations.

- Transformation from existing to next generation architectures to simplify the architectures and better align them with the businesses they enable, and provide the means to externalize and manage policy across all architecture layers.

- Pain points that might be eliminated altogether by migration to next generation architectures.

To satisfy the requirements of next century computing, cloud computing will need to mean more than just externalized data centers and hosting models. Although architectures that we deploy in data centers today should be able to run in a cloud, simply moving them into a cloud stops well short of what one might hope that Cloud Computing will come to mean. In fact, tackling global-scaled collaboration and trading partner network problems in government, military, scientific, and business contexts will require more than what current architectures can readily support. For example:

- It will be necessary to rapidly set up a temporary collaboration network enabling network members to securely interact online, where interaction could imply interoperability with back office systems as well as human oriented exchanges — all in a matter of hours. Examples that come to mind include emergency medical scenarios, global supply chains and other business process networks. Policies defining infrastructure and business constraints will be varied, so policy must be external to, and must interact with, deployed functionality. These examples also imply the need for interoperability between public and private clouds.

- Business interactions have the potential to become more complex than personal transactions. Because they are likely to be formed as composite services, and because services on which they depend may be provisioned in multiple clouds, the ability to provision and uniformly manage composite cloud services will be required, as will be the ability to ensure that these services satisfy specified business policy constraints.

- The way that users and access control are managed in typical applications today is no longer flexible enough to express roles and responsibilities that people will play in next-generation business interactions. Roles will be played by people outside of or across corporate boundaries in an online context just as frequently as they are inside. Access control and the management of roles and responsibilities must be externalized from business functionality so that it becomes more feasible to composite functional behavior into distributed service oriented applications that can be governed by externalized policy.

These considerations suggest that clouds will have to have at least the following characteristics:

- Clouds should be uniquely identifiable so that they can be individually managed even when combined with other clouds. This will be necessary to distinguish and harmonize cloud business and infrastructure policies in force.

- A cloud should be dynamically configurable: configuration should be automatable in varying and unpredictable, possibly even event-driven, conditions.

- Systems management technologies for clouds must integrate constraints on business with constraints on infrastructure to make them manageable in aggregate.

- A cloud should be able to dynamically provision itself and optimize its own construction and resource consumption over time.

- A cloud must be able to recover from routine and extraordinary events that might cause some or all of its parts to malfunction.

- A cloud must be aware of the contexts in which it is used so that cloud contents can behave accordingly. For example, if clouds are composited, policy will have to be harmonized across cloud boundaries; when in multitenant mode, service level agreements may be used to determine priority access to physical resources. Application platforms today are unaware of their usage context, but business functionality in next-generation platforms will have to be managed with context in mind.

- A cloud must be secure, and it must be able to secure itself.

IV. SWITCH FROM TRADITIONAL IT TO THE CLOUD

There are many reasons why organizations of all sizes and types are adopting this model of IT. It provides a way to increase capacity or add capabilities on the fly without investing in new infrastructure, training new personnel, or licensing new software. Ultimately, it can save companies a considerable amount of money.
1. Removal / reduction of capital expenditure
Customers can avoid spending large amounts of capital on purchasing and installing their IT infrastructure or applications by moving to the cloud model. Capital expenditure on IT reduces available working capital for other critical operations and business investments. Cloud computing offers a simple operational expense that is easier to budget for month-by-month, and prevents money being wasted on depreciating assets. Additionally, customers do not need to pay for excess resource capacity in-house to meet fluctuating demand.

2. Reduced administration costs
IT solutions can be deployed extremely quickly and managed, maintained, patched and upgraded remotely by your service provider. Technical support is provided round the clock by reputable providers like ThinkGrid for no extra charge, reducing the burden on IT staff. This means that they are free to focus on business-critical tasks, and businesses can avoid incurring additional manpower and training costs.

3. Improved resource utilisation
Combining resources into large clouds reduces costs and maximizes utilisation by delivering resources only when they are needed. Businesses needn’t worry about over-provisioning for a service whose use does not meet their predictions, or under-provisioning for one that becomes unexpectedly popular. Moving more and more applications, infrastructure, and even support into the cloud can free up precious time, effort and budgets to concentrate on the real job of exploiting technology to improve the mission of the company. It really comes down to making better use of your time – focusing on your business and allowing cloud providers to manage the resources to get you to where you need to go. Sharing computing power among multiple tenants can improve utilisation rates, as servers are not left idle, which can reduce costs significantly while increasing the speed of application development. A side effect of this approach is that computer capacity rises dramatically, as customers do not have to engineer for peak loads.

4. Economies of scale
Cloud computing customers can benefit from the economies of scale enjoyed by providers, who typically use very large-scale data centres operating at much higher efficiency levels, and multi-tenant architecture to share resources between many different customers. This model of IT provision allows them to pass on savings to their customers.

5. Scalability on demand
Scalability and flexibility are highly valuable advantages offered by cloud computing, allowing customers to react quickly to changing IT needs, adding or subtracting capacity and users as and when required and responding to real rather than projected requirements. Even better, because cloud-computing follows a utility model in which service costs are based on actual consumption, you only pay for what you use. Customers benefit from greater elasticity of resources, without paying a premium for large scale.

6. Quick and easy implementation
Without the need to purchase hardware, software licences or implementation services, a company can get its cloud-computing arrangement off the ground in minutes.

7. Helps smaller businesses compete
Historically, there has been a huge disparity between the IT resources available to small businesses and to enterprises. Cloud computing has made it possible for smaller companies to compete on an even playing field with much bigger competitors. ‘Renting’ IT services instead of investing in hardware and software makes them much more affordable, and means that capital can instead be used for other vital projects. Providers like ThinkGrid take enterprise technology and offer SMBs services that would otherwise cost hundreds of thousands of pounds for a low monthly fee.

8. Quality of service
Your selected vendor should offer 24/7 customer support and an immediate response to emergency situations.

9. Guaranteed uptime
Always ask a prospective provider about reliability and guaranteed service levels – ensure your applications and/or services are always online and accessible.

10. Anywhere Access
Cloud-based IT services let you access your applications and data securely from any location via an internet connection. It’s easier to collaborate too; with both the application and the data stored in the cloud, multiple users can work together on the same project, share calendars and contacts etc. It has been pointed out that if your internet connection fails, you will not be able to access your data. However, due to the ‘anywhere access’ nature of the cloud, users can simply connect from a different location – so if your office connection fails and you have no redundancy, you can access your data from home or the nearest Wi-Fi enabled point. Because of this, flexible / remote working is easily enabled, allowing you to cut overheads, meet new working regulations.

11. Technical Support
A good cloud computing provider will offer round the clock technical support. ThinkGrid customers, for instance, are assigned one of our support pods, and all subsequent contact is then handled by the same small group of skilled engineers, who are available 24/7. This type of support model allows a provider to build a better understanding of your business requirements, effectively becoming an extension of your team.

12. Disaster recovery / backup
Recent research has indicated that around 90% of businesses do not have adequate disaster recovery or business continuity plans, leaving them vulnerable to any disruptions that might occur. Providers like ThinkGrid can provide an array of disaster recovery services, from cloud backup (allowing you to store
important files from your desktop or office network within their data centres) to having ready-to-go desktops and services in case your business is hit by problems[17]. Hosted Desktops (or Hosted VDI) from ThinkGrid, for example, mean you don’t have to worry about worry about data backup or disaster recovery, as this is taken care of as part of the service. Files are stored twice at different remote locations to ensure that there's always a copy available 24 hours a day, 7 days per week.

V. CLOUD AS A NEXT GENERATION ARCHITECTURE

The computer architecture of a computing system defines its attributes as seen by the programs that are executed in that system, that is, the conceptual structure and functional behavior of the machine hardware. Then, the computer architect defines the functions to be executed in the hardware and the protocol to be used by the software in order to exploit such functions. Note that the architecture has nothing to do with the organization of the data flow, the logical design, the physical design, and the performance of any particular implementation in the hardware.

Hence By Architecture we mean the order in which certain hardware Processes are carried out by the OS and has nothing to do with the logical software flow of the Computer.

An Operating System is the layer between the hardware and software, as shown in Fig. 3.

<table>
<thead>
<tr>
<th>utilities</th>
<th>applications</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>operating system</td>
<td></td>
</tr>
<tr>
<td></td>
<td></td>
</tr>
<tr>
<td>hardware</td>
<td></td>
</tr>
</tbody>
</table>

Fig. 3. Operating System architecture model

An Operating System is responsible for the following functions

- Device management using device drivers
- Process management using processes and threads
- Inter-process communication
- Memory management
- File systems

In addition, all operating systems come with a set of standard utilities. The utilities allow common tasks to be performed such as

- being able to start and stop processes
- being able to organize the set of available applications
- organize files into sets such as directories
- view files and sets of files
- edit files
- rename, copy, delete files
- communicate between processes

Cloud architecture, [19] the systems architecture of the software systems involved in the delivery of cloud computing, typically involves multiple cloud components communicating with each other over a loose coupling mechanism such as a messaging queue. Elastic provision implies intelligence in the use of tight or loose coupling as applied to mechanisms such as these and others. Cloud computing is the use of computing resources (hardware and software) that are delivered as a service over a network (typically the Internet) [18]. The name comes from the use of a cloud-shaped symbol as an abstraction for the complex infrastructure as shown in figure 4. Cloud computing entrusts remote services with a user's data, software and computation.

Fig. 4: Cloud Computing

CONCLUSION

Cloud computing is the next big wave in computing. It has many benefits, such as better hardware management, since all the computers are the same and run the same hardware. It also provides for better and easier management of data security, since all the data is located on a central server, so administrators can control who has and doesn't have access to the files. In this paper, we presented the introduction of cloud computing and different parameters. It discuss about the feature which plays a big role in switching from existing IT sector to cloud computing. It gives a brief of overview of next generation computers.
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Review on Cloud Computing and Security Issues

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ABSTRACT: Though the concept of cloud is not new, it has undoubtedly proven a major commercial success over recent years. It is a virtual pool of resources that is provided to the users who have to pay for it on demand via internet. Users try to make best use of these services without the knowledge of underlying architecture. In this paper, the types of cloud as well as the deployment of cloud services has been analyzed. Further the emerging issues in cloud computing like data management, privacy and security has been discussed.

Keywords: Cloud computing, Security, Service Models, Information Technology, Infrastructure.

I. INTRODUCTION
Cloud computing is one of the most promising technologies to facilitate development of cost-efficient and flexible on demand computing infrastructure. This rapidly emerging technology prevents the organization from setting up the expensive network architecture to perform large scale and complex computing. Instead of keeping data on a local drive or updating applications for our own needs, we use a service over the internet, to store the information and use its applications. That is in a cloud computing, the entire data resides at multiple locations and is accessed through virtual machines. Since these multiple sites can be located anywhere around the world beyond the reach of users, there are various security and data management issue that needs to be dealt with.

This paper starts by understanding what cloud computing is and further analyzing its underlying framework which comprises of three components. The first component addresses the essential characteristics required for developing the cloud computing model. The second component deals with the service models which are further categorized as Cloud Software as a Service (SaaS), Cloud Platform as a Service (PaaS) and Cloud Infrastructure as a Service (IaaS). And the last component is the Cloud Services which are typically made available via a private cloud, community cloud, public cloud or hybrid cloud. Further in this work-in-progress paper we closely watch out the issues that need to be dealt with like data security, privacy and data management at multiple sites.

II. CLOUD COMPUTING OVERVIEW
The origin of the term cloud is vague. The ‘Cloud’ was first originated from the telephone industry during 1990’s when virtual private network (VPN) service came into existence. Telecommunication companies that had historically only offered single dedicated point-to-point data connections started offering virtualized private network connections with the same service quality as their dedicated services at a reduced cost. Rather than building out physical infrastructure to allow more users to have their own connections, telecommunication companies were able to provide users with shared access to the same physical infrastructure. This change allowed the telecommunication to re-route traffic as necessary to allow for better network balance and more control over bandwidth usage. With the huge network traffic it was difficult to find which route the data would take between the consumer and the provider. Due to this the service provider’s responsibilities were symbolized by the cloud symbol. That’s why the ‘cloud’ in the phrase cloud computing specify the internet and its underlying architecture [2].

Cloud computing is internet based computing where shared resources, applications, and information are provided to the users stored on the provider’s server or computer to the user’s computers or other devices on demand. Cloud computing relies on sharing of resources. Cloud resources are not only usually shared by multiple users but also dynamically allocated for users as per their demand. Cloud services allow individuals and organizations to use software and hardware which are managed by third parties at remote sites. Examples of cloud services include online file storage, social networking sites, webmail and online business applications. Cloud computing provides a shared pool of resources including data storage space, system processing power, networks and specialized corporate and user applications.

III. STRUCTURE OF CLOUD COMPUTING
In a simple, topological sense, cloud computing structure is made up of several elements: clients, the datacenter and distributed servers. As shown in Fig 1 these components make up the three parts of the cloud computing architecture. Each element performs a specific role in delivering resources to the consumer.

Fig1: COMPONENTS OF CLOUD COMPUTING STRUCTURE
A. CLIENTS: Clients in a cloud computing architecture represents the local area network. These are the computers which reside at local site. These can be laptops, tablets, mobile phones etc. all big drivers for cloud computing. Clients are the devices that end users interact with in order to manage their data on the internet i.e. cloud.

B. DATACENTER: The datacenter is the collection of servers where the application we request resides. It can be a collection of servers within an organization or any other site in the world which can be accessed via an internet.

C. DISTRIBUTED SERVERS: Servers are generally not located in the same site. They are dispersed at various locations. It provides more flexibility to the service provider in terms of options and security.

The cloud computing components comprises of various sub components which have their specific function to perform shown in Fig 2.

Fig2: STRUCTURE OF CLOUD COMPUTING

Role of these components have been discussed below:

i.) User Interaction Interface: The first sub component, user interaction interface is how users make request for the services residing on cloud.

ii.) Cloud servers: Cloud servers describe virtual or physical servers managed by system management.

iii.) Monitoring and Meeting: This component tracks the usage of the cloud so the resources used can be attributed to a certain user.

iv.) Service Catalog: It is a list of services that the user can request.

v.) System Management: It is the part which is responsible for managing the available resources.

vi.) Provisioning Tool: carves out the systems from the cloud to deliver on the requested service.

IV. TYPES OF CLOUD SERVICES

Cloud computing provides an environment created in a user’s machine from an online application stored on the cloud and run through a web browser. Cloud computing providers offer their services in terms of fundamental models which are Infrastructure as a Service (IaaS), Platform as a Service (PaaS) and software as a Service (SaaS) [1] [2]. Cloud computing platform can be divided into three categories:

A. Application or Software as a Service (SaaS): SaaS is the most widely known and widely used form of cloud computing. It provides various services to the customers without bothering about application servers, storage, application development and related common areas of IT sector. That is user does not manage or control underlying architecture. Cloud-based applications—or software as a service (SaaS)—run on distant computers “in the cloud” that are owned and operated by others and that connect to users’ computers via the Internet and, usually, a web browser. Examples include applications like Yahoo, Google, Skype.

B. Platform as a Service (PaaS): Platform as a Service model provides a framework to allow developers to build applications and services over the internet. In other words Platform as a service provides a cloud-based environment with services required to support the complete life cycle of building and delivering web-based (cloud) applications—without the cost and complexity of purchasing and managing the underlying hardware, software, provisioning and hosting.

C. Infrastructure as a Service (IaaS): it is the most basic component of cloud computing model. It can be said to be the grass root level of cloud computing. Organizations can build a complete computing infrastructure on demand. Infrastructure as a service provides companies with computing resources including servers, networking, storage, and data centre space on a pay-per-use basis. The consumer does not manage the underlying structure of cloud computing but has full control over the operating systems, storage, networks and deployed applications.

V. DEPLOYEMNT OF CLOUD SERVICES

Any organization can deploy cloud computing in several different ways depending on the factors like where the cloud services are hosted, security requirements, desire to share cloud services, ability to manage or control the services. Different cloud computing deployment models categorized on the basis of its use are explained below: [2][5].

A. Public Cloud: Public clouds allows other organizations/users to access services in fastest possible way and are owned as well as managed by any organization in world.
wide. With public cloud deployment model the customer does not need to purchase hardware, software or supporting infrastructure, which is owned and operated by the providers. The cloud services and cloud resources are obtained from very large resource pools which are shared by all end users. These organizations which are specially built for running cloud computing systems provide the services according to the required quantity.

B. Private Cloud: Private cloud is owned and managed by a single organization which delivers the services to a particular customer or a constituent group within a company. That is, the private cloud infrastructure is operated solely for a single organization. The organization specifies architects and manages a pool of computing resources that the cloud computing service provider delivers as a standardized set of services. A common reason for organizations to procure private clouds is to take advantage of many of cloud’s efficiencies, while providing more control of resources and steering clear of multi-tenancy.

C. Community Cloud: Community cloud is owned and managed jointly by several organizations or communities that share specific needs such as security requirements, policy and compliance considerations. When organizations have a common set of requirements then the community cloud enables them to combine assets and share a pool of resources, data and capabilities.

D. Hybrid Cloud: A hybrid cloud service deployment model uses a private cloud model combined with the strategic use of public cloud services i.e. it implements the required processes by combining the cloud services of different cloud computing models, e.g. private and public cloud services. Hybrid deployment models are complex to manage and operate especially when the communication among models is required and is necessary.

VI. ISSUES IN CLOUD
Cloud computing is the use of computer resources via internet. Although cloud computing can offer services based on pay-as-you-go access method to organizations- the services do comes with certain risks[1][3].

A. Security Issues: Security is the most important issue in cloud. Everything is placed in providers premises which makes the data highly unsecured. Some important security concerns are:

i.) **Secure data transfer:** All the traffic that travels within a network and whatever services the cloud offers must traverse through the internet. In order to deliver the services intact to the customer, data should be travelled from a secure channel because there is a possibility where a malicious user can corrupt data. Users may tolerate rare service interruption but non recoverable data losses can kill a business.

ii.) **Secure software interfaces:** The interfaces that are used to interact with the cloud services should be strong. Relying on weak software interfaces exposes organizations with various security concerns like availability, confidentiality and integrity.

iii.) **Secure stored data:** The data that is to be delivered to the customer should be encrypted when it’s on the provider’s server and while it’s in use by the cloud service.

iv.) **User access control:** Data stored on a cloud provider’s server should be accessible only to the intended clients within a company. Also, the sensitivity of the data which is being allowed outside the company should be considered carefully.

<table>
<thead>
<tr>
<th>SECURITY CONCERNS</th>
<th>SOLUTION FOR SECURITY CONCERN</th>
<th>PROBLEM</th>
</tr>
</thead>
<tbody>
<tr>
<td>Who controls the encryption/decryption keys?</td>
<td>Service provider using strong encryption algorithm</td>
<td>Encryption can make data totally unusable and complicates the usability</td>
</tr>
<tr>
<td>Data Loss</td>
<td>Backup</td>
<td>Increases load on server</td>
</tr>
<tr>
<td>Network security</td>
<td>User can deny the access of any internet based service by using IP spoofing.</td>
<td>Can cause security harm</td>
</tr>
</tbody>
</table>

Table 1: COMPARATIVE STUDY OF SECURITY ISSUES IN CLOUD COMPUTING

B. Data issues: deals with the problems that is encountered while transferring data to the user [6]. The main issues can be:

i.) **Data loss:** Data loss is a very serious problem in cloud computing. If the vendor does not provide services any more due to any financial or legal problems there will be a chance of data loss to the customers. The customer won’t be able to access it due to unavailability of data.

ii.) **Data integrity:** When it comes to location of data nothing is transparent even the customer is not aware about where his data is located. The provider never reveals the exact location of data. Data integrity is also an important issue in cloud data storages because it ensures completeness, correctness, accessibility, consistency and high quality data. The data integrity is
a mechanism of writing the data in a reliable manner to the persistent data storages such that it can be retrieved in the same format when requested. In cloud, the complete storage of data is done at data centers/ data storages and the security and integrity of data lies on the vendor storing data in the data centers. Cloud storage is becoming an important issue for the outsourcing of day to day management of data. Therefore integrity monitoring of data in cloud is important for data centers, to avoid any kind of data corruption or data crash. Data failure can occur at any storage level. One of the most famous data failure occurred in Amazon which led to complete data loss stored in it. Just storing the data at cloud data centers does not ensure integrity of data, but some mechanism should be implemented at each storage center that ensures data integrity.

iii.) Data confidentiality: Current cloud offerings are generally public networks (rather than private networks) exposed to more attacks.

C. Cost Issues: Using cloud can significantly reduce the cost of installing software and required hardware but the cost of traversing data over the network increases. Cloud and the cost per unit of services being used is likely to be higher in cloud computing. This problem is particularly prominent when hybrid cloud computing model is used where the organization’s data is distributed amongst a number of public/private/community clouds.

CONCLUSION

Cloud computing has the potential to lead in promoting a promising, secure and economically feasible IT solution in the future. It helps the organizations to get efficient use of their IT hardware and software resources at low cost. In this paper we have firstly discussed the architecture of cloud computing along with the types of services offered by it. The next section highlights the deployment models of Cloud Computing. This paper also analyzes various security considerations that are currently being faced in cloud computing. It also emphasizes on the security objectives that need to be achieved. Cloud computing has massive prospects, but the security, data management and privacy threats encountered are directly proportion to its offered advantages. The future of cloud is really alluring, giving the vision of communication at low cost. At present, the focus in cloud computing is on mesh architecture and large scale.

REFERENCES

A Review of Cost Model for Object serialization using XML and JSON formatters

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ABSTRACT
This paper presents Object Serialization Techniques that can be useful for various purposes, including object serialization. Minimization, which can be used to decrease the size of serialized data. We have implemented means by serialization and de-serialization of object can be done using modern format XML and JSON after adding compression or encryption or possibly both to the object streams.

Keywords: Object Serialization, Compression Techniques, Object oriented design, Performance Analytics.

1. Introduction
Serialization is the process of converting complex objects into a stream of bytes for storage. De-serialization is its reverse process that is unpacking stream of bytes to their original form. It is also known as Pickling, the process of creating a serialized representation of object.

The following steps are necessary to do to create a serializable class:
1. Create a custom class with assigned properties.
2. Define the serialization functions.
3. Create a Controller class and instantiate our Custom class.
4. Serialize the object to a named file.
5. De-serialize the values by reading it from the file.

Object serialization has been investigated for many years in the context of many different distributed systems.

Most popular Serialization formats
There are various data serialization formats available for developers according to choose form. There are also various ways to convert complex objects to sequences of bits. It does not include markup languages used exclusively as document file formats.
- Binary Format Serialization
- XML Format Serialization
- XML-RPC Serialization
- JSON Serialization
- YAML[C] Serialization

The following are the basic advantages of serialization: First is to facilitate the transportation of an object through a network and secondly create a clone of an object that can be restored later on.

2 Related Work

In the paper “Object Serialization and De-serialization Using XML”¹, inter operability of potentially heterogeneous databases has been an ongoing research issue for a number of years in the database community. With the trend towards globalisation of data location and data access and the consequent requirement for the coexistence of new data stores with legacy systems, the cooperation and data interchange between data repositories has become increasingly important. The emergence of the Extensible Markup Language (XML) as a database independent representation for data offers a suitable mechanism for transporting data between repositories.

This paper describes a research activity within a group at CERN (called CMS) towards identifying and implementing database serialization and deserialization methods that can be used to replicate or migrate objects across the network between CERN and worldwide centers using XML to serialize the contents of multiple objects resident in object oriented databases.

The paper “Generic Pickling and Minimization”² presents generic pickling and minimization mechanisms that are provided as services similar to garbage collection. Pickling is used to externalize and internalize data. Minimization means to maximize the sharing in arbitrary data structures. The paper introduces the notion of an abstract store as a formal basis for the algorithms, and analyzes design decisions for the implementation aspects of pickling and minimization. The mechanisms presented here are fully implemented in the Alice programming system.

We presented a generic pickling and minimization mechanism. We showed how Alice, as a conservative extension of Standard ML, uses pickling in a type safe way for its component system. To build a formal base for the algorithms, we introduced abstract stores as a universal memory model. Un-pickling and pickling are based on this model, allowing us to analyze and evaluate our design decisions such as bottom up versus top down un-pickling and right to left traversal. Minimization can be used to decrease the size of pickled data. However, the general mechanism presented here seems suitable for other applications such as efficient representation of runtime types. Finally, we extended the system with support for concurrency as present in Alice. The authors analyzed how pickle and minimize must behave in such a concurrent setting.

In the paper “Why Object Serialization is Inappropriate for Providing Persistence in Java”³, the author paper describes why Object Serialization is not appropriate for providing...
Persistence in Java. With numerous code examples, Object Serialization is shown to be easy to work with initially which seduces the developer into relying on it for persistence within more complex applications.

The advanced use of object serialization requires significant work from the programmer, something that is not apparent at first. The use of object serialization together with static and transient fields and within multithreaded programs is discussed together with the "big inhale problem": the need to read in the entire object graph before processing over it can commence.

This paper has shown, with numerous supporting examples, that using Java's object serialization mechanism to provide object persistence is inappropriate. The system appears simple on the surface but there are many implications from relying on it as a persistence technology. The programmer must state the types that are candidates for persistence at compile time, whereas making this decision at runtime, on a per object basis, is more appropriate.

The serialization mechanism suffers from the big inhale problem where the whole graph must be read before it can be used; loading objects on demand is more efficient, reducing delay in starting an application. The serialization mechanism creates copies of objects that it writes and reads. This can break some code that makes assumptions about the hash code of an object.

<table>
<thead>
<tr>
<th>µs per object</th>
<th>32 int w</th>
<th>4 int, 2 null w</th>
<th>tree(15) w</th>
<th>r</th>
<th>r</th>
<th>w</th>
<th>r</th>
</tr>
</thead>
<tbody>
<tr>
<td>JDK serialization</td>
<td>346 1410</td>
<td>169 506 1192 1889</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>UKA-serialization</td>
<td>35 30 19 28 201 354</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>improvement %</td>
<td>90 97 89 95 83 81</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>explicit marshaling</td>
<td>228 920 81 308 396 647</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>slim type encoding</td>
<td>19 187 16 159 72 213</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>internal buffering</td>
<td>0 139 6 19 0 330</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>buffer accessibility</td>
<td>48 65 30 49 502 291</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>two types of reset</td>
<td>16 40 17 33 21 54</td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

Fig 1: Object Serialization for various type of Objects with Encodings\[5\].

The complexity of using object serialization within la distributed environment, when evolving classes and when using specialized class loaders is also discussed. The paper compares the performance of serializing and de-serializing a byte array and binary tree of the same data size to and from an NFS mounted disk and two kinds of local disk. Alternative solutions to object persistence in Java are presented at the end of the paper.

Using Experiments carried out by author draws four conclusions:

1. The absolute amount of time to read and write a store is large;
2. Reading a store is much slower than writing a store; and if an application is likely to exhibit more reading than writing,
3. an NFS mounted disk should be used;
4. The use of JIT technology significantly increases the speed of using Java object serialization.

In the "Object Serialization in the .NET Framework"\[6\], the author describes using Serialization in .Net framework. He describes the two most important reasons are to persist the state of an object to a storage medium so an exact copy can be recreated at a later stage, and to send the object by value from one application domain to another.

It is also used by remoting to pass objects by value from one application domain to another. This paper provides an overview of the serialization used in the Microsoft .NET Framework.

The author gives Serialization Guidelines; one should consider serialization when designing new classes since a class cannot be made serializable after it has been compiled. Some questions to ask are: Do one have to send this class across application domains? Will this class ever be used with remoting? What will users do with this class? Maybe they derive a new class that needs to be serialized. When in doubt, mark the class as serializable. It is probably better to mark all classes as serializable unless:

- They will never cross an application domain. If serialization is not required and the class needs to cross an application domain, derive the class from MarshalByRefObject.
- The class stores special pointers that are only applicable to the current instance of the class. If a class contains unmanaged memory or file handles, for example, ensure these fields are marked as Non-Serialized or don't serialize the class at all.

Comparison between JSON and YAML for data serialization\[7\], this report determines and discusses the primary differences between two different serialization formats, namely YAML and JSON. A general introduction to the concepts of serialization and parsing is provided first, which also explains how they can be used to transfer and store data. This is followed by an analysis of the YAML and JSON formats, where functionality, primary use cases, and syntax is described. In addition to this the perceived performance of implementations for both formats will also be investigated by conducting a number of tests.

Using the combined background information and results from the tests, conclusions regarding the main differences between the two are then determined and discussed.

As has been concluded, it is clearly very easy to read thanks to the required usage of whitespace and the ability to skip surrounding quotes for strings. YAML also has the advantage of allowing comments in the document. Users can easily read and manipulate the output, which is one of the reasons as to why it’s often used for configuration files.

This enables the straightforward definition of strongly-typed objects that match serialized structures, for example existing XML formats. Inheritable translation scopes group sets of object serialization binding definitions, and enable inheritance. The present system supports (compressed) XML for...
serialization, while future work will develop alternate translation schemes, such as type-length-value and JSON.

Execution times in seconds for serialization.

<table>
<thead>
<tr>
<th>Method</th>
<th>Simple</th>
<th>Complex</th>
</tr>
</thead>
<tbody>
<tr>
<td>JSON.generate</td>
<td>0.1550s</td>
<td>0.5830s</td>
</tr>
<tr>
<td>JSON.pretty_generate</td>
<td>0.1470s</td>
<td>0.6060s</td>
</tr>
<tr>
<td>YAML.dump</td>
<td>2.4531s</td>
<td>3.4732s</td>
</tr>
</tbody>
</table>

Execution time in seconds for De-Serialization.

<table>
<thead>
<tr>
<th>Method</th>
<th>Simple</th>
<th>Complex</th>
</tr>
</thead>
<tbody>
<tr>
<td>JSON.parse</td>
<td>0.0440s</td>
<td>0.0790s</td>
</tr>
<tr>
<td>YAML.load</td>
<td>0.2750s</td>
<td>0.3360s</td>
</tr>
</tbody>
</table>

Table 2: JSON VS. YAML Serialization performance

The execution times measured for the serialization/deserialization process shows their results, similar to the serialization process, which can be seen in table 2. Both implementations are much faster at generating data structures from a serialized string than doing the opposite. YAML is also slower.

CONCLUSION

The primary design goals for Serialization, to provide a simple and effective data exchange, but also being easy to generate and load. It is widely used and is used natively available in the most common modern programming. Object Serialization as presented here is especially well suited for functional programming languages, where the closure semantics and the ability to serialize code is essential. Also a minimization technique helps reduce Serialization sizes considerably.

FUTURE SCOPE

To implement means by which Serialization and Deserialization of Objects can be done using modern formats XML and JSON after adding Compression or Encryption or possibly both to the Object Streams.

In future I also want to see how the Performance of Object Serialization is affected in a Normal CLR Binary VS. Native JIT compiled Binary. The perceived performance of XML or JSON can be determined from a custom benchmarking. A complex data set will also be used to test performance of the implementations for documents with deeper hierarchies.

Another direction for future work may be to modify the semantics of the serialization algorithm to improve performance.
Review: Log mining in Databases

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Abstract — In last decade lots of research was done on log mining. In fact log mining became an interesting area of research. This paper analyzes traditional log mining. Log mining is a part of web mining and web mining in turn is a part of data mining which a part is itself of Knowledge Discovery in Databases (KDD). In this paper it has been rewied that how log mining will be efficient in field of databases.

I. INTRODUCTION

Knowledge Discovery in Databases creates the context for developing the tools needed to control the flood of data facing organizations that depend on ever-growing databases of business, manufacturing, scientific, and personal information.

Web mining is the application of data mining techniques to discover patterns from the Web. Discovering useful information from the World-Wide Web and its usage patterns. The application of data mining techniques to extract knowledge from Web data. Web data is Web content i.e. text, image, records, and etc. Web structure contains hyperlinks, tags, etc. Web usage uses http logs, app server logs, etc. Current software application often some auxiliary text files known as log files. Such files are used during various stages of software development, mainly for debugging and profiling purposes. Use of log files helps testing by making debugging easier. It allows following the logic of the program, at high level, without having to run it in debug mode. Nowadays, log files are commonly used also at customer’s installations for the purpose of permanent software monitoring and/or _n-tuning. Log files became a standard part of large application and are essential in operating systems, computer networks and distributed systems. Log files are often the only way how to identify and locate an error in software, because log file analysis is not affected by any time-based issues known as probe effect. Log files are often very large and can have complex structure. Although the process of generating log files is quite simple and straightforward, log _le analysis could be a tremendous task that requires enormous computational resources, long time and sophisticated procedures.

II. TRADITIONAL WEB LOG MINING

Web log mining is the process of applying data mining technologies to discover usage patterns from the Web data. One important source to discover such patterns is the Web log data that contains users Web browsing history. Web Usage Mining addresses the problem of extracting behavioral patterns from one or more web access log. The entire process can be divided into three major steps. The first step, pre-processing, is the task of accurately identifying pages accessed by web visitors. This is a very difficult task because of page caching and accesses by web crawlers. The second step, pattern discovery, involves applications of data mining algorithms to the pre-processed data to discover patterns. The last step, pattern analysis, involves analysis of patterns discovered to judge their interestingness.
Web server records all users’ activities of the web site as web servers Logs. Most log files have text format and each log entry is saved as a line of text. There are many types of web logs such as NCSA format, W3C format and IIS format, but they share the same basic information. These log data can be used in web site designing, modifying and also to improve the overall performance of web site. After identifying the different web server log data files there is a need to merge the log files.

K. R. Suneetha and Dr. R. Krishnamoorthi[2] have analyzed NASA server log file of size 195MB, various analysis has been carried out to identify the user behavior. The errors which arise in Web surfing were determined. Grace. L. K. Joshi; Maheswari, V.; Nagamalai, Dhinahar[3] determine that Log files contain information about User Name, IP Address, Time Stamp, Access Request, number of Bytes Transferred, Result Status, URL that Referred and User Agent. The log files are maintained by the web servers. By analyzing these log files gives a neat idea about the user. This paper gives a detailed discussion about these log files, their formats, their creation, access procedures, their uses, various algorithms used and the additional parameters that can be used in the log files which in turn give way to an effective mining. It also provides the idea of creating an extended log file and learning the user behavior.

In 2010 Mr. Singh B and Mr. Singh H.K has done research in the area of web mining. The aim of this research is to provide past, current evaluation and update in each of the web context (6) Web usage mining [7] is referred to the discovery of user access patterns from web usage logs, which records every click made by the users. This information is frequently gathered and automatically stored into access logs through Web server.

Web usage mining process is similar to data mining process. The difference is in data collection phase. The data are collected from databases for data mining whereas it is collected from web log files in web usage mining. In conventional data mining techniques information pre-process includes data cleaning, integration, transformation and reduction. But web mining pre-processing categorize into Content pre-processing, Structure pre-processing, Usage pre-processing. Once the data is collected from log files, a three-step process is performed in web usage mining namely data preparation, pattern discovery and pattern analysis.

Maheswara Rao [8] introduced a new framework to separate human user and search engine access intelligently with less time span. And also Data Cleaning, User Identification, Sessionization and Path Completion are designed correctly. The framework reduces the error rate and improves significant learning performance of the algorithm. Perkowitz et al.

III. APPLICATION OF LOG MINING

Each of the applications can benefit from patterns that are ranked by subjective interesting.

Web usage mining is used in the following areas:

a. Web usage mining offers users the ability to analyze massive volumes of click stream or click flow data, integrate the data seamlessly with transaction and demographic data from offline sources and apply sophisticated analytics.

b. Personalization for a user can be achieved by keeping track of previously accessed pages. These pages can be used to identify the typical browsing behavior of a user and subsequently to predict desired pages. By determining frequent access behavior for users, needed links can be identified to improve the overall performance of future accesses.

c. In addition to modifications to the linkage structure, identifying common access behaviors can be used to improve the actual design of Web pages and to make other modifications to the site.

d. Web usage patterns can be used to gather business intelligence to improve Customer attraction, Customer retention, sales, marketing and advertisement, cross sales.
VI. PROS AND CONS OF WEB LOG MINING

A. Pros
Web usage mining essentially has many advantages which makes this technology attractive to corporations including the government agencies. This technology has enabled e-commerce to do personalized marketing, which eventually results in higher trade volumes. Government agencies are using this technology to classify threats and fight against terrorism. The predicting capability of mining applications can benefit society by identifying criminal activities. The companies can establish better customer relationship by giving them exactly what they need. Companies can understand the needs of the customer better and they can react to customer needs faster. The companies can find, attract and retain customers; they can save on production costs by utilizing the acquired insight of customer requirements. They can increase profitability by target pricing based on the profiles created. They can even find the customer who might default to a competitor the company will try to retain the customer by providing promotional offers to the specific customer, thus reducing the risk of losing a customer or customers.

B. Cons
Web usage mining by itself does not create issues, but this technology when used on data of personal nature might cause concerns. The most criticized ethical issue involving web usage mining is the invasion of privacy. Privacy is considered lost when information concerning an individual is obtained, used, or disseminated, especially if this occurs without their knowledge or consent. The obtained data will be analyzed, and clustered to form profiles; the data will be made anonymous before clustering so that there are no personal profiles. Thus these applications de-individualize the users by judging them by their mouse clicks. De-individualization, can be defined as a tendency of judging and treating people on the basis of group characteristics instead of on their own individual characteristics and merits. Another important concern is that the companies collecting the data for a specific purpose might use the data for a totally different purpose, and this essentially violates the user’s interests. The growing trend of selling personal data as a commodity encourages website owners to trade personal data obtained from their site. This trend has increased the amount of data being captured and traded increasing the likelihood of one’s privacy being invaded. The companies which buy the data are obliged make it anonymous and these companies are considered authors of any specific release of mining patterns. They are legally responsible for the contents of the release; any inaccuracies in the release will result in serious lawsuits, but there is no law preventing them from trading the data. Some mining algorithms might use controversial attributes like sex, race, religion, or sexual orientation to categorize individuals. These practices might be against the anti-discrimination legislation. The applications make it hard to identify the use of such controversial attributes, and there is no strong rule against the usage of such algorithms with such attributes. This process could result in denial of service or a privilege to an individual based on his race, religion or sexual orientation, right now this situation can be avoided by the high ethical standards maintained by the data mining company. The collected data is being made anonymous so that, the obtained data and the obtained patterns cannot be traced back to an individual. It might look as if this poses no threat to one’s privacy, actually many extra information can be inferred by the application by combining two separate unscrupulous data from the user.

CONCLUSION
In this paper, we try to find out method for log mining. In today’s world million of people accessing the web at same time. In billion of entry are save in the log on web. These logs become wider day by day and this effect the accessing of web. Many of the log information are not of our need like when we access a web page then due to some error they not open. Lots of research has on this log. We find that in each paper they firstly make crystal of this information. Most of author use k mean algorithm for that. After that different author use different algorithm for find frequent set like aprior, SOM etc. These algorithms are very effective log mining.

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Energy Management: A challenging factor in cloud computing

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Abstract: Cloud computing has emerged as a new paradigm that is based on pay as you go model and delivers resource services like CPU, storage, application, services etc to the users anywhere anytime on demand through internet connectivity. Several large companies like Yahoo, Amazon, IBM are trying hard enough to provide powerful, reliable and cost efficient cloud platform to serve to end users and get best possible benefit from this paradigm. Data centres hosting cloud applications consume large amount of energy in form of electricity leading to rising cost. Increasing power consumption has become a concern for cloud systems. Therefore, it is necessary to increase the efficiency of data centre at minimum cost. Virtualization of data centre enables the efficient operation of servers by server consolidation and migration strategies. In fact, cloud computing provides so many features like no extra investment on infrastructure required, minimum operating cost, scalability, easy access [1] but still there are some issues and challenges that need to be looked upon like virtual machine migration, automated service provisioning, server consolidation and energy management. In this paper, various challenges, issues and techniques used to minimize the energy consumption in cloud computing have been focused.

Keywords: Cloud computing, Energy management, Virtualization, Energy efficiency, SLA.

I. INTRODUCTION

Cloud computing is a model that provides persistent, convenient, on demand network access to a shared pool of configurable computing resources (e.g., networks, servers, storage, application and services) that can be rapidly provisioned and released with minimal management effort or service provider interaction. The characteristics of cloud computing include on-demand self service, broad network access, resource pooling, rapid elasticity and measured service. The cloud computing service models are Software as a Service (SaaS), Platform as a Service (PaaS) and Infrastructure as a Service (IaaS). In Software as a service model a complete application is offered to the customer, as a service on demand i.e. a pre-made application, along with any required software, operating system, hardware, and network are provided. In PaaS, an operating system, hardware, and network are provided, and the customer installs or develops its own software and applications. The IaaS model provides just the hardware and network; the customer installs or develops its own operating systems, software and applications.

NIST definition of cloud computing:

“Cloud computing is a model for enabling convenient, on-demand network access to a shared pool of configurable computing resources (e.g., networks, servers, storage, applications, and services) that can be rapidly provisioned and released with minimal management effort or service provider interaction”[4].

Rapid development of storage and processing technologies has enabled the upcoming trends of computing model called cloud computing in which resources are leased and released by the users through internet in on-demand basis. It has been estimated that the cost of powering and cooling accounts for 53% of the total operational expenditure of data centres. In 2006, data centres in the US consumed more than 1.5% of the total energy generated in that year, and the percentage is projected to grow 18% annually. Hence infrastructure providers are under enormous pressure to reduce energy consumption [1]. Based on a recent ‘Data Centre Energy Forecast Report’, it can be expected that savings of the order of 20% can be achieved in server and network energy consumption with respect to current levels, and that these savings may induce an additional 30% saving in cooling needs as detailed in a study by HP and the Uptime Institute. It shows that ‘most of data centre power is spent on cooling ICT equipment (between 60 and 70%). Thus there are very significant economic and environmental gains to be obtained from a serious research thrust on energy efficiency in the general area of IT and computer networks [2].

Fig1. Energy distribution in data centre [2]

High energy consumption in data centres results in increased electricity bills, additional requirement of cooling infrastructure e.g. UPS or PDU’s. Apart from these irresistible operating and acquisition costs, high energy dissipation effects environmental balance too as large amount of CO₂ is emitted. With the growth
and increased demand of cloud computing there will be more data transfer and storage requirement. Hence energy management in cloud computing is great area of concern and various techniques should be used to minimize it in best possible way. Earlier power usage in cloud systems was done by using Fine-grain and Coarse-grain methods. In fine-grain methods power sensors are directly connected to hardware devices and power reading is recorded whereas in coarse-grain technique power of group of servers is traced by external power meters. But for large scale cloud systems, power management is done at VM’s as it is just impractical to connect thousands of servers to the power meters.

In cloud systems delivering IaaS, dedicated VM’s and resources are allocated to the users and green computing is practiced by the data centres for efficient utilization of resources. Virtualization is the key feature of cloud computing that allows efficient utilization of resources. Scalability allows end users to up-scale or down-scale the resources in a cloud depending upon their requirement. To achieve energy management in such cases two kinds of virtual machine migration techniques are used. Regular migration moves a VM from one host to another by pause and resume procedure whereas second process also does the same thing without need of suspending the server. This technique is called Live Migration. Many migration schemes have been proposed to get efficient cloud system. Live Migration means that the virtual machine seems to be responsive every time during the migration process [3]. Benefits of live migration is load balancing, energy saving and online maintenance. Virtual machine migration technique is one of the most effective ways of applying virtualization. Another novel approach for energy conservation is server consolidation, means, running several multiple servers on a single server. It allows physical servers to be turned off via migrating the virtual machines to other servers. Now, request sent by the user to the cloud is passed on to the data centre via broker. Broker forwards this request to the scheduler and scheduler schedules the load distribution to the data centre in such way that load balancing is maintained.

II. Cloud Components

The main components with specific functionalities that make a cloud are: Client, data centre and distributed servers.

Clients: Clients are the ubiquitous devices like laptops, mobiles, tablet computers that just sits and asks for the resources and services to the cloud via. Internet. These can be Mobile phones, thin clients that do not have internal hard drive and totally depends on servers for all operations and thick clients which is normal computers that we use in day to day life.

Data Centre: It is central system of a cloud that provides the provision for storage, networking equipments, cooling equipments computations and contains thousands of devices like switch, router and servers. All the applications and services requested by the user are serviced by the data centre. Therefore, it is very necessary to properly schedule each request for network performance and throughput so that energy consumption is also maintained.

Virtual Machines: Virtualization is done through the virtual machines which allow the proper utilization of resources. Virtual machine is the software implementation that provides remote environment to the application to install and execute. The figure is an indication of virtual machine diagram in which multiple number of virtual machines work on one physical host. Virtual machine monitor is a software implementation of a program it translates the request from the virtual machine into physical host and allocates the CPU, memory, and hardware [5].

![Virtualized machine diagram](image)

Distributed Server: Servers distributed around various geographic locations are called distributed servers. This provides more security and number of options to access the data and services. For example, Amazon has its cloud solution at various places. If anything goes wrong in any of the server, data can be still accessed via other servers.

III. Migration techniques

To improve the reliability of virtual machine system memory migration or file system migration is done. Two types of migration techniques are, Pre-copy and Post-copy. In pre-copy bulk memory is migrated to the destination node without stopping the execution of virtual machine continues while its
migrated. To ensure consistency, pages modified during this phase are re-sent. Post copy migration includes transfer of memory content after the process state is transferred to destination [6]. In post-copy approach memory pages are transferred at-most once.

Main goals of migration technique are:

**Load Balancing:** It prevents some machines from getting fully occupied whereas others having spare space. Therefore, migration allows the equal distribution of load among the virtual machines.

**Server Consolidation:** It allows multiple VM’s to be migrated and executed on single physical server to maintain the resource utilization so that unused virtual machines can be turned off. It results in reduced power consumption and reduced operational cost.

**Balanced resource utilization:** These are detected by the VM owner based on the threshold set by data centre owner based on SLA (Service Level Agreement) with the end user. Machines using resources above threshold value form the hotspot and signifies over-utilization of resource whereas vice-versa form the coldspot and signifies under-utilization of resources.

**A. Live Migration**

This technique allows the migration of one physical server to other server without disrupting the services or connection. It results in very small downtime approximately in milliseconds. Downtime is the time duration during which service remains unavailable to the user as virtual machine has no current request to be processed.

**B. Virtual Machine Consolidation**

Flexibility in having on demand infrastructure cloud computing has led to the rising Total Cost of Ownership. But several companies are striving lot to reduce the cost of energy, infrastructure, power and supply. Virtual Machine consolidation maximizes the resource utilization while minimizing the energy consumption by increasing the number of inactive machines and minimizing the number of active machines to save energy. Since the difference in energy consumption between an idle and a suspended server is quite high, suspending an inactive server provides another prospect to cut the energy consumption. Server consolidation can be done in two ways. First way is to move large number of applications from dedicated servers in a non-virtualized cluster to a small number of high performance virtualized servers. In second type large virtualized data centres where applications are wrapped within virtual machines (VMs) and mapped on demand and released frequently .Here VM mapping are changed by dynamically reallocating VMs [7] without stopping their execution.

**IV. Power Management Techniques for Energy Saving**

Power management of a data centre can be done by using hardware, software or firmware. We can also use OS for provisioning proper system utilization. Sometimes while performing virtualization hardware resources get over sized, therefore during average utilization the system remains idle and can be used to save the power. Several technologies used to reduce the energy consumption are:

- **A. Power capping:** Used in data centres having power constraints where availability of power is limited. In this method, less power and cooling facilities are provided to the system thus reduces the extra overhead of infrastructure cost. During average resource utilization, power consumption remains less and cannot exceed the maximum threshold value. If server exceeds the power cap threshold value, the system gets blocked in order to maintain the system performance.

- **B. Dynamic Voltage and Frequency Scaling (DVFS):** In this technique, clock is used to track the resource utilization of the system. Clock frequency of the processor is decreased when there is minimum utilization of resources. The set of operating points for the processor is abstracted called as P-states or Processor Performance States. P-states are the operational, optimal state combination of voltage and frequency is abstracted based on the workload demands. Higher the P-state, less frequency as well as voltage would be required by the processor to operate. Higher P-state represents the slow operational speed hence saving the power. The OS further apply policies to meet the SLA with end user [8].

- **C. CPU idle power saving states:** Modern processors have several power saving modes called C-states. C-state represents the capability of a processor to turn off its unused components to save power. It has basically four states C0, C1, C2, and C3. In C0 state instructions are executed. Other C-states represent the processor is idle and higher the C-state value, deeper it would be in sleep mode. As deep the processor is in sleep mode, more power is saved. But drawback, of deep sleep mode is that processor have slower wake up time and can be resolved by the trade-off by the OS through some heuristics and policies.

**DESIGNING OPTIMAL ENERGY POLICY:** In this paper [9], author has discussed various policies based on three factors namely “Performance”, “Balancing” and Powersave”. ‘Performance’ profile focuses on getting maximum throughput with minimum latency. In ‘Balanced’ profile, energy management techniques are applied. When the workload is increased, system should provide extra boost in performance and when the system is idle, system must shut down to save the power. ‘Powersave’ profile is mostly used in low priority jobs especially when difference between idle power consumption and average runtime is not too much. In such cases, it is better to throttle the system and run it slower by using ‘powersave’ profile [9].

The experimental result of these profiles depicting power and performance characteristics with varying system utilization has been shown by the author[9].
CONCLUSION

In recent years, energy management has become an important requirement for modern computing systems as large amount of energy is being consumed and dissipated by the data centre. With the rapid growth of cloud computing and its scalability, energy consumed by the data centre is directly proportional to the number of hosted servers and the total workload. This also raises the CO₂ level in an environment. Therefore, to restrict this rapidly growing energy consumption some efficient techniques and algorithms need to be designed. In this paper, survey on some of the energy saving techniques like live migration has been explored.

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Service Quality of Metro Railways in Delhi - A Brief Study of Delhi and NCR.

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ABSTRACT
In today’s competitive state, consumer satisfaction is the first main concern of any organization for which they should aim not only at satisfying the customer but also focus on the delighting them. So it becomes nuts and bolts for organization to identify the factors that affect customer satisfaction level and consciously measure them so as to try and bring about the necessary changes on the basis of customer perception and requirements. This present research using data collected through a structured questionnaire from a sample of 500 respondents tries to find the factors related to Delhi Metro Railway service excellence that have an impact on customer satisfaction. Service quality has been viewed as a determinant of customer satisfaction in today competitive market scenario. To maintain cordial relationship with the customer, organization needs to understand and meet the expectations of its customers. The organization at the moment should aim not only at satisfying the customer but should focus on delighting them by providing qualitative services. This study identifies components of service quality of Delhi Metro Railways. The study is exploratory in nature and uses factor analysis of research to identify the most important factors of customer satisfaction with service quality. The study was conducted using survey method. The findings reveal that out of the various factors considered, inconvenience due to construction has maximum effect on satisfaction level of customers with Delhi Metro as a whole.

Keywords: service quality, customer satisfaction, Delhi Metro Railway, exploratory research

1. INTRODUCTION
The Delhi Metro is a rapid transit system serving Delhi, Gurgaon, Noida and Ghaziabad in the National Capital Region of India. The network consists of six lines with a total length of 189.63 km with 142 stations of which 35 are underground, five are at-grade and rest are elevated.

Planning for the metro started in 1984, when the Delhi Development Authority(DDA) and the Urban Arts Commission came up with a proposal for developing a multi-modal transport system for the city. The Government of India and the Government of Delhi jointly set up the Delhi Metro Rail Corporation (DMRC) in 1995. Construction started in the 1998, and the first section, on the Red Line, opened in
2. BACKGROUND OF DELHI METRO RAILWAYS

The concept of a mass rapid transit (MRT) for New Delhi first emerged from traffic and travel characteristics study which was carried out in the city in 1969. Over the next several years; many official committees by a variety of government departments were commissioned to examine issues related to technology, route alignment and governmental jurisdiction. In 1984, the Delhi Development Authority and the Urban Arts Commission came up with a proposal for developing a multi-modal transport system, which would consist of constructing three underground mass rapid transit corridors as well as augmenting the city’s existing suburban railway and road transport networks.

While extensive technical studies and the raising of finance for the project were in progress, the city expanded significantly resulting in a twofold rise in population and a fivefold rise in the number of vehicles between 1981 and 1998. Consequently, traffic congestion and pollution soared, as an increasing number of commuters took to private vehicles with the existing bus system unable to bear the load.

An attempt at privatizing the bus transport system in 1992 merely complicated the problem, with inexperienced operators plying poorly maintained, noisy and polluting buses on lengthy routes, resulting in long waiting times, unreliable service, extreme overcrowding, unqualified drivers, speeding and reckless driving. To rectify the situation, the Government of India and the Government of Delhi jointly set up a company called the Delhi Metro Rail Corporation (DMRC) on March 5, 1995 with E. Sreedharan as the managing director. Dr. E. Sreedharan handed over the charge as MD, DMRC to Shri Mangu Singh on 31 December 2011.

After the previous problems experienced by the Kolkata Metro, which was badly delayed and 12 times over budget due to “political interfering, technical troubles and bureaucratic delays”, the DMRC was given full powers to hire people, decide on tenders and control funds. Construction work the Delhi Metro started on October 1, 1998 then consulted the Hong Kong MTR on rapid transit operation and construction techniques. As a result, construction proceeded smoothly, except for one major disagreement in 2000, where the Ministry of Railways forced the system to use broad gauge despite the DMRC’s preference for standard gauge.

The first line of the Delhi Metro was inaugurated by Atal Behari Vajpayee, the then Prime Minister of India on December 24, 2002 and thus it became the second underground rapid transit system in India, after the Kolkata Metro. The first phase of the project was completed in 2006 on budget and almost three years ahead of schedule, an achievement described by Business Week as “nothing short of a miracle”.

3. REVIEW OF THE LITERATURE

Various scholars have considered different dimensions of service quality.

Gronoos (1884) considers technical, functional, and reputation quality; Lehtinen and Lehtinen (1982) consider interactive, physical, and corporate quality; and Hedvall and Palschik (1989) focus on willingness and ability to serve and the physical and psychological access to the service. In conceptualizing the basic service quality model, Parasuraman (1985) identify 10 key determinants of service quality as perceived by the service provider and the consumer, namely, reliability, responsiveness, competence, access, courtesy, communication, credibility, security, understanding/knowing the customer, and tangibility to formulate a service.
quality framework. Later (in 1988), they modify the framework to five determinants: reliability, assurance, tangibles, empathy, and responsiveness. The techniques of customer satisfaction analysis allow the critical aspects of the supplied services to be identified and customer satisfaction to be increased (Cuomo 2000). The literature also shows that researchers have identified different factors of quality in the context of different services. Vanniarajan and Stephen (2008) identify the attributes that passengers use to evaluate the service quality of Indian Railways as reliability, assurance, empathy, tangibles, and responsiveness. It is found that passengers are “moderately satisfied” to “satisfied” on these dimensions.

Agrawal (2008) identifies employees’ behavior as most important determinant of customer (passenger) satisfaction with Indian Railway services. Eboli and Mazzulla (2007) measure customer satisfaction in the context of bus service on various factors including availability of shelter and benches at bus stops, cleanliness, overcrowding, information system, safety, personnel security, helpfulness of personnel, and physical condition of bus stops. J. D. Power and Associates (2008) also measures customer satisfaction with high-speed and dialup Internet service providers based on five factors: performance and reliability cost of service, customer service, billing, and offerings and promotions.

In a study on Internet banking it has been found that consumers give the highest weight to the quality of service while selecting a particular bank (Geetika et al. 2008). In another study of customer satisfaction with banking services, it appears that factors of customer satisfaction are traditional (basic) facilities, convenience, behavior of employees, and the environment of bank (Jham and Khan 2008).

Some of the earliest studies in this area were undertaken by Allen and DiCesare (1976) who considered that quality of service for the public transport industry contained two categories: user and non-user. The user category consists of speed, reliability, comfort, convenience, safety, special service and innovation. Another study named service quality attributes affecting customer satisfaction for bus transit for measuring the relationship between global customer satisfaction and service attributes of public transport especially of bus transit for university of Calabria students to reach the campus from the urban area of Cosenza of southern Italy.

A model proposed in this study which may useful to analyze the correlation between service quality attributes and identify the more convenient attributes for improving the supplies service (Fu, L. and Xin, Y. 2007). This study provides the methodological assistance to conduct current study to determine the relationship between rail passenger satisfaction and service attributes. Especially multivariate technique, factor analysis, regression analysis and analysis of variance were used to estimate the interrelated dependency of attributes. In current study basically factor analysis and regression analysis used to draw the relationship between the satisfaction of service and service quality attributes of rail passenger satisfaction and service attributes. Specifically multivariate technique, factor analysis, regression analysis and analysis of variance were used to estimate the interrelated dependency of attributes.

Eboli and Mazzulla (2007) measured customer satisfaction of bus on various factors including availability of shelter and benches at bus stops, cleanliness, overcrowding, information system, safety, personnel security, helpfulness of personnel, and physical condition of bus stops. TCRP report identifies the following elements at bus station for efficient service: shelters, waiting rooms and seating, doorways, stairways, escalators, signage and information display, public address system, and passengers’ amenities. In a study on internet banking, consumer gave the highest weight to the quality of service while selecting a particular bank (Geetika et al. 2008).

In another study of customer satisfaction with banking services, factors of consumer satisfaction were basic facilities, convenience, behaviors of employees, and the environment of bank (Jham and Khan 2008). Customer satisfaction with full-service moving companies was measured across seven factors: transportation of belongings, loading service, unloading service, optional coverage, estimate process, packing service and insurance/damage claims. This implies that the quality of basic facilities and other supporting facilities were used as criteria for satisfaction (J.D.Power and Associates Reports 2007).

Trippa and Drea (2002) also used a survey of Amtrak passengers to assess the ‘direct and indirect relationship between pre-core/ peripheral and core service performance components and their impact on the likelihood of repeat purchase’. They found that the core experience on-board determined the customers’ attitude to the service provider and subsequently their intention to use the train again. These attributes included announcements, seat comfort, and ride, cleanliness of the seating area, courtesy of on-board staff, rest rooms and cloak room.

4. OBJECTIVE OF THE RESEARCH

This study primarily aims to assess the effects of consumer perceptions of the various aspects of services provided by public and private transportation services on their level of satisfaction with reference to Delhi Metro Railways.

The study aims to propose a framework of the major dimensions that have an impact on the perceived quality of the services provided by the Delhi Metro Railways and hence their overall satisfaction.

So the objective of the study is as follows-

a) To find the importance attached to the various aspect of Delhi Metro by the customers.

b) To identify the factors affecting customer satisfaction with Delhi Metro Railways based on customer’s perception of the quality of performance of these factors.
c) To analyze direction and magnitude of the effects of these factors on customers’ overall satisfaction with Delhi Metro Railways.

3) Promotional design: Promotional design offered by the Metro Railway includes Smart card, Minimum Multi Ride, Limited Multi ride and Extended Multi Ride.

(3): There is no significant relationship between promotional design and customer satisfaction.

5) Personal Interaction: personal interaction includes Staff Behavior, Attitude of the Management, Helpfulness of Staff and Attentiveness and resolve Quarries of customer.

(4): There is no significant relationship between Personal Interaction and customer satisfaction.

6. RESEARCH METHODOLOGY

On the basis of literature review, the researchers conducted a consumer survey. A total of 500 respondents were strategically selected to conduct the pilot survey in different Metro stations in Delhi, Ghaziabad and Noida (Table 1). A reliability test has been done. Resulting Cronbach Alpha (Coefficient Alpha) for all the factors indicated an acceptable level of reliability.

The study has been conducted to identify the factors that mostly satisfy the customer. Another reason of the pilot study is to refine the test instrument. On the basis of factors identified in the pilot study stage, a structured questionnaire has been constructed on Likert 5 point scale to conduct a market survey. The questionnaire includes questions on each of the variables for measuring customer satisfaction.

Thus, a questionnaire has been used as survey instrument for conducting the survey. Data has been generated by face to face interview. Respondents have been asked to give their perception of the quality level across the different factors as well as satisfaction level toward their Metro services in Delhi, Ghaziabad and Noida.

The survey encompasses evaluation from eight different Metro stations in Delhi, Ghaziabad and Noida. A brief profile of sample respondents is presented in table- 1. The customer satisfaction relating to Delhi Metro Railway services are measured by using the Chi-square test of 12 degree of freedom at 5% significant level.

### TABLE-I : CLASSIFICATION OF RESPONDENTS ACCORDING TO THEIR PROFILE

<table>
<thead>
<tr>
<th>Sl. No.</th>
<th>Factor</th>
<th>Category</th>
<th>No. of respondents</th>
<th>%</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Gender</td>
<td>Male</td>
<td>324</td>
<td>64.8</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Female</td>
<td>176</td>
<td>35.2</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Total</td>
<td>500</td>
<td>100</td>
</tr>
<tr>
<td>2</td>
<td>Age</td>
<td>Below 25 years</td>
<td>94</td>
<td>18.8</td>
</tr>
<tr>
<td></td>
<td></td>
<td>26-35 years</td>
<td>152</td>
<td>30.4</td>
</tr>
<tr>
<td></td>
<td></td>
<td>36-45 years</td>
<td>124</td>
<td>24.8</td>
</tr>
<tr>
<td></td>
<td></td>
<td>46-55 years</td>
<td>86</td>
<td>17.2</td>
</tr>
<tr>
<td></td>
<td></td>
<td>Above 55 years</td>
<td>44</td>
<td>8.8</td>
</tr>
</tbody>
</table>
The table-1 indicates that:

a). 64.8% respondents are male and rest of 35.2 % is female respondents.

b). 18.8% respondents’ fall in the age group up to 25 years, 30.4% respondents were from age group of between 25-35 years, 24.8% and 17.2 % respondents were from age group between 36-45 years and 46-55 years respectively and only 8.8% respondents were above 55 years.

c). In reference to level of education, 11.2% respondents were educated up to school level, 21.2% respondents were Higher Secondary level, 44.8% respondents were graduate level and remaining 18.8% respondents were postgraduate.

d). 16.4% respondents were engaged in government sector, 30.4% respondents were belongs to private sector, 24.8% respondents performs own business, 15.6% respondents were students and rest of the respondents i.e., 12.8% were fell under the household category.

e). 16.8% respondents belonged to the income bracket of below Rs. 10000, 25.6% respondents were from income group of Rs. 10001-15000, 31.6% and 15.2% respondents were belonged to income level of Rs. 15001-25000 and 25001-25000 respectively and only 10.8% respondents fell under the income group of above Rs. 25000.

To sum up, it can be concluded that most of the respondents avail the Delhi Metro Railway were male. Out of Metro rail passengers most were belonged to 25-35 years age group. 44.8% respondents were graduates, majority of respondents were from private sector job holders and majority of the respondents were belongs to middle class (Rs.15001-25000) income level.

7. ASSOCIATION TEST TO EXPLORE THE COMPONENT OF CONSUMER SATISFACTION

<table>
<thead>
<tr>
<th>Service characteristics</th>
<th>Level of customer satisfaction</th>
<th>Total Score (%)</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>Service characteristics</td>
<td>Highly Satisfied</td>
<td>Satisfied</td>
<td>Neutral</td>
</tr>
<tr>
<td>Ticketing</td>
<td>248</td>
<td>224</td>
<td>14</td>
</tr>
<tr>
<td>Arrival &amp; departure Inf.</td>
<td>204</td>
<td>238</td>
<td>24</td>
</tr>
<tr>
<td>Security</td>
<td>232</td>
<td>196</td>
<td>34</td>
</tr>
<tr>
<td>Life &amp; escalator</td>
<td>224</td>
<td>212</td>
<td>36</td>
</tr>
</tbody>
</table>

TABLE-II: SERVICE CHARACTERISTIC AND LEVEL OF CUSTOMER SATISFACTION
The result (Table-2) from chi-square test indicates that Ho(1) is rejected, because the chi-square value is 26.46, which is higher than the table value (21.026) at the level of significance 0.05. Thus alternative hypothesis is accepted which implies that the service characteristics are significantly related with the customer satisfaction. Table (2) gives an overall view of the customer satisfaction with the sub-dimensions of the service quality. Ticketing got the 1st position with the score of 587 and percentage (27.56%) as the service characteristic of Delhi Metro Railway followed by the lift and escalator system (24.98%). Arrival and departure information stands on third position with 24.18%. Security got the percentage (23.28%). It is reasonable to assume that except the security, ticketing, lift and escalator and arrival and departure information satisfy most of customers.

### TABLE-III: PHYSICAL ASPECT AND LEVEL OF CUSTOMER SATISFACTION

<table>
<thead>
<tr>
<th>Dimensions</th>
<th>Highly Satisfied</th>
<th>Satisfied</th>
<th>Neutral</th>
<th>Dissatisfied</th>
<th>Highly Dissatisfied</th>
<th>Total Score (%)</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>Air condition &amp; lighting</td>
<td>234</td>
<td>216</td>
<td>12</td>
<td>30</td>
<td>8</td>
<td>531(25.23)</td>
<td>2nd</td>
</tr>
<tr>
<td>Seating and free space</td>
<td>220</td>
<td>228</td>
<td>24</td>
<td>8</td>
<td>20</td>
<td>504(23.94)</td>
<td>4th</td>
</tr>
<tr>
<td>Inside atmosphere</td>
<td>238</td>
<td>214</td>
<td>18</td>
<td>24</td>
<td>6</td>
<td>547(25.49)</td>
<td>1st</td>
</tr>
<tr>
<td>Parking space outside</td>
<td>222</td>
<td>230</td>
<td>8</td>
<td>12</td>
<td>28</td>
<td>523(24.48)</td>
<td>3rd</td>
</tr>
</tbody>
</table>

The above result (Table-3) from the chi-square test indicates that the physical aspect is significantly related with the customer satisfaction as the chi-square value is 27.04 which is higher than chi-square table value (21.026) at the level of significance 0.05. So, Ho (2) is rejected. Thus H1 is accepted. The table (3) shows an overall view of the customer satisfaction with this physical aspect. Inside atmosphere got the highest score 547 and percentage 25.49% followed by the Air conditioning facility (25.23%). Parking Space outside and seating and free space scores respectively 24.48% and 23.94%. According to the result it can be concluded that most of the features of physical aspect meet the customer expectation while there is a need to focus on the increase of seating capacity for senior citizens and ladies.

### TABLE-IV: PROMOTIONAL SCHEME AND LEVEL OF CUSTOMER SATISFACTION

<table>
<thead>
<tr>
<th>Dimension</th>
<th>Highly Satisfied</th>
<th>Satisfied</th>
<th>Neutral</th>
<th>Dissatisfied</th>
<th>Highly Dissatisfied</th>
<th>Total Score (%)</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>Smart card</td>
<td>238</td>
<td>196</td>
<td>16</td>
<td>32</td>
<td>18</td>
<td>522(25.16)</td>
<td>3rd</td>
</tr>
<tr>
<td>Minimum Multi Ride</td>
<td>218</td>
<td>238</td>
<td>12</td>
<td>10</td>
<td>34</td>
<td>523(25.20)</td>
<td>2nd</td>
</tr>
<tr>
<td>Limited Multi Ride</td>
<td>228</td>
<td>214</td>
<td>32</td>
<td>14</td>
<td>12</td>
<td>498(24.0)</td>
<td>4th</td>
</tr>
<tr>
<td>Extended Multi Ride</td>
<td>256</td>
<td>198</td>
<td>18</td>
<td>20</td>
<td>8</td>
<td>532(25.64)</td>
<td>1st</td>
</tr>
</tbody>
</table>

From the above table result from the chi-square test we can conclude that as Ho is rejected, the promotional scheme and the level of customer satisfaction are significantly related as the chi-square value are 22.15 which are higher than the table value 21.026 at the level of significance 0.05. As the table shows that Extended Multi Ride (EMR)-Valid for 90 days (80 Rides journey available by paying fare for 55 Rides) is the most significant factor that affects the customers...
satisfaction, followed by Minimum Multi Ride (MMR) - Valid for 21 days (12 Rides journey available by paying fare for 11 Rides). It can conclude that promotional schemes meet some of respondents’ expectation. There is a need to focus on the promotional schemes, especially on the

Limited Multi Rides in terms of increase the some number of rides which cover the total journey period in a month.

### TABLE-5: PERSONAL INTERACTION AND LEVEL OF CUSTOMER SATISFACTION

<table>
<thead>
<tr>
<th>Dimension</th>
<th>Highly Satisfied</th>
<th>Satisfied</th>
<th>Neutral</th>
<th>Dissatisfied</th>
<th>Highly Dissatisfied</th>
<th>Total Score (%)</th>
<th>Rank</th>
</tr>
</thead>
<tbody>
<tr>
<td>Behavior of Staff</td>
<td>232</td>
<td>224</td>
<td>8</td>
<td>26</td>
<td>10</td>
<td>532(25.49)</td>
<td>2&lt;sup&gt;nd&lt;/sup&gt;</td>
</tr>
<tr>
<td>Management Attitude</td>
<td>220</td>
<td>232</td>
<td>28</td>
<td>10</td>
<td>20</td>
<td>535(25.63)</td>
<td>1&lt;sup&gt;st&lt;/sup&gt;</td>
</tr>
<tr>
<td>Helpfulness of Staff</td>
<td>198</td>
<td>240</td>
<td>12</td>
<td>34</td>
<td>16</td>
<td>490(23.48)</td>
<td>4&lt;sup&gt;th&lt;/sup&gt;</td>
</tr>
<tr>
<td>Attentiveness &amp; resolve Quarries</td>
<td>246</td>
<td>202</td>
<td>22</td>
<td>22</td>
<td>8</td>
<td>530(25.40)</td>
<td>3&lt;sup&gt;rd&lt;/sup&gt;</td>
</tr>
</tbody>
</table>

Chi-square test (Table-5) reveals that personal interaction is significant related with the customer satisfaction as the chi-square value is 21.31 which is higher than the table value 21.026. It indicates that Ho is rejected. The overall picture shows that most of the respondents are satisfied and significant numbers of respondents are highly satisfied with the personal interaction.

### 8. FINDINGS

Based on the analysis and interpretation of data collected the following are the findings can be summarized as follows:

a) The male passengers avail the Metro services outnumber of female passengers with the ultimately goal of cost minimization and saving the time.

b) Majority of respondents travel by Metro rail belongs to the age group between 26-35 years.

From the results of Chi-square test it is clear that:

c) Majority of respondents avail the Metro as mode of transportation for ticketing i.e., promotional offer.

d) Inside atmosphere attract the customer for select Metro as a mode of communication neglecting the other obstacles.

e) Extended Multi Ride scheme helps to secure the cost minimum journey.

f) Attitude of the management is also play a key role to increase the customer satisfaction.

### 9. RECOMMENDATIONS

Based on the findings, the following suggestions may be recommended to improve the Metro rail service in Delhi.

a) As the majority of the respondents avail the Metro services for reaching the ultimate destination of office, so it should be need smoother and better system of ticketing and information regarding the arrival and departure of trail. It is also important to improve the quality of security at the time of journey to reduce pick pocket and others.

b) Senior respondents and ladies passengers are also attract by the Metro services and they use the Metro service regularly that’s why it is necessary to arrange or increase the number of seat inside the station.

c) Lift & escalator, Inside atmosphere, reasonable cost are the major motivating factors that influence the customer to avail the Metro services, so the Metro authority should give emphasis on these factors.

d) Proper parking facilities should be there and parking should be made by minimum rent for regular customers. Such people may be issued a parking card.

e) Special attention should be need for promotional scheme of limited multi ride which create a gap of offer journey and actual journey in a month.

### 10. LIMITATIONS OF THE STUDY

1) As sample size is 500, it is not necessary that it truly represents the population/ universe.
2) Some people might not give accurate responses, which affects the results of the study.

3) Some respondents have not taken the schedule seriously and hence, the researcher had to discard those responses.

4) The data collected from Delhi, Ghaziabad and Noida were not same and there was a significant differences among them.

11. SCOPE OF FURTHER RESEARCH

1) Sample size should be increased as no of commuters are increasing day by day in addition to that no of platform are also increasing.

2) The data collected from Uttar Pradesh, Haryana and Delhi should be compared because it will give very wide disparity among them.

12. CONCLUSION

Although several studies have attempted to identify factors determining over all user satisfaction with Indian Railways but this study is the identification of factors that determine passengers’ satisfaction with the quality of services provided by Delhi Metro.

The study is based on empirical research where determinants identified are Convenience, Services on Platform/Train Reservation Counters, Employee behavior, Availability of Trains. With increase of Metro demand in the routes, excessive pressure on Delhi Metro Rail service has emerged. But accordingly increasing demand of service no satisfactory imitative have been performed simultaneously.

However in most case the existing service quality has not observed at satisfactory provision. From the satisfaction model it have observed that the satisfaction of metros service is depend on five distinct service quality attributes and convenience is the worst among those because the coefficient of the convenience get high value which implies the service satisfaction is mostly dominated by this factor.

Delhi Metro Railways needs to attend to the satisfaction of their passengers, particularly with respect of these variables considered importantly by customers in their evaluation and choice. The study thus provided a direction for Delhi Metro Railway administration whereby areas for improving services may be identified and passengers’ satisfaction may be enhanced.

References


Abstract------Breaking encrypted passwords has been of interest to hackers for a long time, and protecting them has always been one of the biggest security problems operating systems have faced, with Microsoft’s Windows being no exception. Due to errors in the design of the password encryption scheme, especially in the Lan Man (LM) scheme, Windows has a bad track in this field of information security. Especially in the last couple of years, where the outdated DES encryption algorithm that LanMan is based on faced more and more processing power in the average household, combined with ever increasing harddisk size, made it crystal clear that LanMan nowadays is not just outdated, but even antiquated. This paper surveys various flaws in window NT environment security.

Keywords:- Cryptography, Window Operating Systems, Rainbow Tables, Hash generation, Algorithms.

VI. INTRODUCTION
LAN Manager [1], or LM, is an authentication protocol designed (at its time) to maximize password security in a Windows-based environment. The LM protocol was first used in Microsoft's LAN Manager Product a very long time ago and is still the authentication protocol of choice for older operating systems, such as Windows 95 and Windows NT 3.51 and earlier. Later, when Windows NT was introduced, LM was enhanced and renamed the NTLM [2] authentication protocol. Although NTLM has been around for a long time, it's still a basically good authentication protocol, and it is the native network authentication protocol of Windows NT 4.0 and earlier operating systems.

A. NTLM MAJOR Weaknesses[3]

• SAM has several vulnerabilities, which allowed attackers to access the hashed passwords.

NTLM can use a maximum of 14 characters to create its stored hash. These 14 characters are split into two seven-character strings. Crypto-graphically, it is reasonably easy to brute force attack[4] two seven-character strings with modern computers.

• NTLM cannot use lowercase letters. It converts all lowercase letters to uppercase before creating the hash. This reduces the character set for the password, making brute force attacks far more likely to succeed.

• The hash algorithm used to store passwords became well known. That allowed attackers to guess users' passwords by running password guesses through the hash until the result matched the result stored in the SAM. Because the algorithm remained constant, large libraries of hashed passwords could be stored and used to quickly attack a SAM.

• NTLM used a mechanism known as pass-through authentication to distribute the authentication task. The way pass-through authentication was designed created a bottleneck at the primary domain controller (PDC) of each domain. Some of the tasks done by the PDC, such as password changes, could not be offloaded to any other server.

• Attackers began accessing passwords by pretending to be trusted servers. Users' client computers would transmit logon information to the attackers, thinking that they were domain controllers or file servers. NTLM provided no way for users to verify that the server they were connecting to be the one they intended to connect to.

• NTLM was largely limited to interoperability with Microsoft products. As computer networks became more heterogeneous, NTLM didn't provide a way to interoperate with non-Microsoft operating systems.

• NTLM provided no way for a middle-tier application to access resources on a user's behalf. When a user's client application accessed a middle-tier application, the middle-tier application usually used a generic
administrator credential to access backend resources. This technique works, but presents a security threat, because the middle-tier application is running under powerful security credentials.

VII.  RAINBOW TABLES

A rainbow table [5] is a way of doing cryptanalysis very quickly and efficiently. Suppose that you are a hacker and you have acquired a database of usernames and encrypted passwords. The System encodes the password using a hash function, which is basically a way of condensing a given set of data into a condensed string. For example, the MD5 algorithm encrypts password “My Password” as 48503dfd58720bd5f-f35c102065a52d7 if one, as a hacker, have the password described above, one wouldn’t know what the password is just by looking.

Instead, one would refer to the rainbow table for the password.

Rainbow tables are a pre-computation based approach to reversing hashes. They require a large amount of pre-computation, but can store the results of this in a reasonable amount of space. When searching for a hash, additional computation is required, but the computation required for searching is significantly less than the amount required for the pre-computation, and significantly less than the amount required to brute force [3] a password.

By generating long chains of passwords and hashes, tied together by the hash function and a reduction function, rainbow tables store a compressed representation of a password search space. By performing similar computations on a provided hash, they are able to dramatically reduce the amount of computation required to find the original password. As with many algorithms, there are limitations with rainbow tables. Unlike a brute force algorithm, they are not guaranteed to find a password within the search space, as the algorithm is probabilistic in the coverage of the password space, and a password will only be found if it is represented in the generated tables. However, very high success probabilities can be achieved, and the search time is significantly less than with a brute force algorithm.

The crack time/storage space tradeoff of rainbow tables is adjusted by changing the chain length. Longer chains require less storage space, but require more computation (and more time) to crack passwords.

VIII.  RELATED WORKS

MARTIN E. HELLMAN, [4], in “A Cryptanalytic Time-Memory Trade-Off”, describes that a probabilistic method is presented which crypt analyzes any N key cryptosystem in N\(^{2/3}\) operations with N\(^{2/3}\) words of memory (average values) after a precomputation which requires N operations. If the precomputation can be performed in a reasonable time period (e.g. several years), the additional computation required recovering each key compares very favorably with the N operations required by an exhaustive search and the N words of memory required by table lookup.

When applied to the Data Encryption Standard (DES) used in block mode. It Indicate that solutions should cost between $1 and $100 each. The method Works in a chosen plain text attack and, if cipher block chaining is not used, can also be used in a cipher text-only attack.

The time-memory trade-off was described for use with a block cipher, but the same approach works with a synchronous stream cipher. The first k bits of key stream are taken as the f(K) function, where K is the number of bits of key. This can be done under a known plaintext attack. The method works on all systems in a chosen plaintext attack but does not work with a known plaintext attack on a cipher feedback system if the initial load of the shift register is random and varies between conversations.

Proposed Federal standards suggest this precaution. Even a block cipher can foil the time-memory trade-off in a known plaintext attack through cipher block chaining or other techniques which introduce memory into the encipherment. Then, even when eight blanks occur in the plaintext, their encipherment depends on the preceding text. Even if the first block of text is fairly standard (e.g., “Login: “), this technique can be foiled by the transmission of a random “indicator” which is used to affect the encipherment (e.g., it is taken as the 0th plaintext block). Again, proposed standards include provision for cipher block chaining with a random indicator. While this time-memory trade-off cryptanalytic technique can be easily foiled, it does work on the DES in basic block mode, more importantly; it indicates that even when cipher block chaining or other techniques are added, a larger key size is needed to have a reasonable assurance of security.

While table lookup and exhaustive search are currently infeasible on systems with 64-bit or larger key sizes, an N\(^{1/2}\) time-memory trade-off would push the minimum usable key size up to 128 bits. The N\(^{2/3}\) technique described here, coupled with the large number of N\(^{1/2}\) time-memory tradeoffs known for other searching problems, indicates that valuable data should not be entrusted to a device with smaller key size.
Philippe Oechslin, [5], in “Making a Faster Cryptanalytic Time-Memory Trade Off”, describes that in 1980 Martin Hellman described a cryptanalytic time-memory trade-off which reduces the time of cryptanalysis by using precalculated data stored in memory. This technique was improved by Rivest before 1982 with the introduction of distinguished points which drastically reduces the number of memory lookups during cryptanalysis. This improved technique has been studied extensively but no new optimizations have been published ever since. The Authors proposed a new way of precalculating the data which reduces by two the number of calculations needed during cryptanalysis. Moreover, since the method does not make use of distinguished points, it reduces the overhead due to the variable chain length, which again significantly reduces the number of calculations. As an example, the authors have implemented an attack on MS-Windows password hashes. Using 1.4GB of data Attacker can crack 99.9% of all alphabet numerical passwords hashes (237) in 13.6 seconds whereas it takes 101 seconds with the current approach using distinguished points. The Authors showed that the gain could be even much higher depending on the parameters used and they have introduced a new way of generating precomputed data in Hellman’s original cryptanalytic time-memory trade-off. Our optimization has the same property as the use of distinguished points, namely that it reduces the number of table look-ups by a factor which is equal to the length of the chains. For an equivalent success rate our method reduces the number of calculations needed for cryptanalysis by a factor of two against the original method and by an even more important factor (12 in our experiment) against distinguished points. The Authors showed that the reason for this extra gain is the variable length of chains that are delimited by distinguished points which results in more false alarms and more overhead per false alarm. They conjecture that with different parameters the gain could be even much larger than the factor of 12 found in our experiment. These facts make our method a very attractive replacement for the original method improved with distinguished points.

The fact that their method yields chains that have a constant length also greatly simplifies the analysis of the method as compared to variable length chains using distinguished points. It also avoids the extra precalculation effort which occurs when variable length chains have to be discarded because they have an inappropriate length or contain a loop. Constant length could even prove to be advantageous for hardware implementations.

Finally their experiment has demonstrated that the time-memory trade-off allows anybody owning a modern personal computer to break cryptographic systems which were believed to be secure when implemented years ago and which are still in use today. This goes to demonstrate the importance of phasing out old cryptographic systems when better systems exist to replace them. In particular, since memory has the same importance as processing speed for this type of attack, typical workstations benefit doubly from the progress of technology.

Hans Hedbom, et al [6], in “A Comparison of the Security of Windows NT and UNIX”, describes that This paper presents a brief comparison of two operating systems, Windows NT and UNIX. The comparison covers two different aspects. First, we compare the main security features of the two operating systems and then we make a comparison of a selection of vulnerabilities most of which we know have been used for making real intrusions.

The Authors found that Windows NT has slightly more rigorous security features than “standard” UNIX but the two systems display similar vulnerabilities. The conclusion is that there are no significant differences in the “real” level of security between these systems.

This paper demonstrates that the security mechanisms of Windows NT are slightly better than those of UNIX. Despite this fact the two systems display a similar set of vulnerabilities. This implies that Windows NT has the theoretical capacity of being more secure than “standard” UNIX. However, with the present way of installing and using the system there seems to be no significant difference between their security levels. It is true that there are presently more intrusions in UNIX systems, but the authors believed that this is due to the aging factor, i.e. the statement above should hold when comparing the systems at the same state of development and market penetration.

Thus, the only reason for more UNIX penetrations is that the system is older and more well-known and we should anticipate an increasing number of intrusions into Windows NT, a tendency that has already started. It is clear that the Achilles heel of both systems is networking. Since both systems utilize the same low level protocols, i.e. IP, TCP and UDP, and comparable high level protocols. This could to some extent explain that the security behavior of both systems is similar, but it does not provide the full explanation.
However, as long as the networking is such a weak point, the usefulness of other security mechanisms is diminished.

Jorgen Blakstad, et al [7], in “All in a day's work: Password cracking for the rest of us”, Describes that the majority of computer systems are still protected primarily with a Username and password, and many users employ the same password on multiple systems. Additionally, some of the most popular operating systems such as Windows XP, Windows Vista and the upcoming Windows 8, still use ad-hoc constructed hash functions such as LM, while many Linux variants use the hash function MD5.

This paper describes an experiment where we have tested the strength of a selection of passwords when converted to LM, NT and MD5 hashes, respectively, using commonly available tools. Our conclusion is that a large number of passwords can be cracked within a normal working day, and that all LM hash passwords can be recovered easily. The use of such weak hash functions in the process of user authentication in these operating systems poses a significant threat to an organization's security.

IX. CONCLUSION

The main benefit of Rainbow Tables is that while the actual creation of the rainbow tables takes much more time than cracking a single hash, after they are generated you can use the tables over and over again. Once you have generated the Rainbow Tables, Attacks using is faster than brute force attacks and needs less memory than full dictionary attacks.

In this paper we have reviewed some of the most important works in rainbow table generation and using rainbow tables in window NT environment, i.e. against NTLM. We have discuss how NTLM is weak against rainbow table attacks.

X. FUTURE SCOPE

Rainbow Tables are popular with a particularly weak password algorithms such as Microsoft LM and NTLM hash, these password algorithm was used in earlier days of Windows and still lives on only for compatibility reasons. In the future We want to devise an experiment where we will test the strength of a selection of passwords using commonly available tools. This is to show that a large number of passwords can be cracked within working days and Majority of passwords used commonly have very skewed frequency distributions. We want to device methodologies based upon calculation of frequency distribution algorithm.

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Comparative Study of Educational ERP System

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Abstract: Enterprise Resource Planning (also known as ERP) is an effective approach that most businesses implement to enhance their productivity and performance. Before implementing this system, it is necessary that business owners and board members have an extensive look at the benefits and risks associated with the use of the ERP system. Known as a systematic approach that most industries use to organize resources as well as improve efficiency and performance, Enterprise Resource Planning (also known as ERP) is usually implemented by corporations to centralize the databases and functions of every department in a single system. The system features various components including software modules, which integrate and manage all the business and private records of firms. With the proper use of this system, firms can decrease their losses and increase their profits.

Keywords: ERP, eCampus, IEMS, EIMS, OLE.

I. INTRODUCTION

ERP systems experienced rapid growth in the 1990s because the year 2000 problem and introduction of the euro disrupted legacy systems. Many companies took this opportunity to replace such systems with ERP. ERP systems initially focused on automating back office functions that did not directly affect customers and the general public. Front office functions such as customer relationship management (CRM) dealt directly with customers, or e-business systems such as e-commerce, e-government, e-telecom, and e-finance, or supplier relationship management (SRM) became integrated later, when the Internet simplified communicating with external parties.[citation needed] “ERP II” was coined in the early 2000s. It describes web-based software that allows both employees and partners (such as suppliers and customers) real-time access to the systems. The role of ERP II expands from the resource optimization and transaction processing of traditional ERP to leveraging the information involving those resources in the enterprise’s efforts to collaborate with other enterprises, not just to conduct e-commerce buying and selling. ERP II is more flexible than the first generation ERP. Rather than confine ERP system capabilities within the organization, it goes beyond the corporate walls to interact with other systems. Enterprise application suite is an alternate name for such systems.

II. LIST OF EDUCATIONAL ERP SYSTEM

1) Educational Management System: Educational Management System is an integrated solution for complete computerization for educational institutions, build on the most futuristic and highly sophisticated Java” environment, denoted as E I M S - Educational Institutes Management System. It is Multilingual ERP Software. The solution has been implemented in many prominent and reputed educational institutions of all levels from multi-branch Nursery Schools, Graded Schools to Colleges of the country. Since, this an Integrated, user configurable and dynamic software solution, it help institutions to get the wide range detailed and summarized information of Administrative and Academic nature, in different forms required at different level of the Organizational hierarchy and for other interested parties like Students, Parents and other Organizations. Educational Institute Management System (EIMS) is best software for schools, Colleges, Institutes, Engineering Colleges, management Colleges, medical Colleges, Nursery, best software for institutions in India, Delhi, college software, Engineering college software, play school software, SMS enabled school software, school software with SMS, IVRS, GPRS and web portal, online school software, software for degree colleges with modules like student admission, registration, attendance, icard, library, result software, software for payroll, income tax, EIMS School Software, CBSE analysis software for school, board result analysis software for school in India, no 1 software for school in India, school software company, sms software for school, IVRS software for school

1.1 EIMS Concepts
1. As an extension of the concept of E I M S software solution, the information stored in centralized server can be shared with the Parents and other organizations related to the Institution through a

2. a) Dynamic Web Portal
   b) Telephone using Interactive Voice Response System (IVRS)
   d) Interactive SMS (Short Message Service) to Mobile Phone
   e) Smart Card
   f) Access Control System
   g) GPRS

![EIMS Concept Diagram](image)

**EIMS Features**

The unique features of E I M S that make it an attractive proposition for users in the utilities sector is:

- Only basic knowledge of computers is required for operation of E I M S. As it has user-friendly application interface.
  
  a) E I M S is **Customizable and User Configurable**.
  
  b) An inbuilt Settings module makes E I M S flexibility to cater to diverse organizational needs.
  
  c) It is build on JAVA technology - one of the most latest and upcoming technologies in the field of Information Technology, which makes you a forerunner in the world of Information technology.
  
  d) E I M S brings information to the user's desktop through integration across all modules.
  
  e) E I M S is Multi-user system. E I M S has administrator controlled user privileges for Multi-level system access control for security.
  
  f) Better co-ordination between student – teacher, parent – institute, teacher- teacher and teacher - management increases quality and effectiveness of education and institution and increased transparency of operations at all levels.
  
  g) E I M S has pre-defined reports. These are used for normal reporting as well as Student & teacher development purpose.
  
  h) E I M S can be easily customized for their own customized reports.

2) **My School ERP: A Complete Solution for Multi-School Management:**

The Education industry is on edge of a radical change. The need of manpower is increasing with the growth in the Education industry, and a huge demand-supply gap is expected in the education space. To overcome these challenges resulting from such gaps, this industry needs IT solutions to manage its resources with optimal efficiency. A ‘S-a-a-S’ based educational ERP system for the schools is a complete School Management Software which helps to upgrade the standard of any school not only in the management level but also helps in transforming the educational what a data warehouse is by comparing these two kinds of systems

3) **Integrated E-educational ERP Solution:**

FPII are a part of entire lifecycle of Universities students and institutes through our E-educational ERP system. Integrated E-educational ERP Solution uses N tier architecture which uses high end web and database servers with load balancing and rescaling. This is compatible with all devices i.e., browsers and mobiles, etc. This ERP is successfully implemented for years in Maharashtra State Board of Technical Education, Mumbai. FPII has developed E-educational ERP to enable the universities to automate its complete process right from its Institute Affiliation to final Result Display of examinations. The University is Admin having its own login, further having logins for Regional Offices, Institutes & Candidates as well.
a) Online Affiliation of Institutes.
b) Online upload of entire Information related to institute.
c) Upload of Affiliation Documents, Courses Information by Institute.
d) Affiliation Information confirmation by Regional Offices.
e) Online search Institute module for Affiliated Institutes.
f) Search by Inst code and Name.
g) Search by Course type.
h) Search by district.
i) Search by Institute Type.

Online Filling & Confirmation of Candidate Enrollment
a) Online Registration by Candidate, Institute or Regional Office.
b) Online upload of Candidates Photo with Signature.
c) Online confirmation of Enrollment Registration form by Institute.
d) Online confirmation of Enrollment registration form by Regional Office.
e) Online generation of Enrollment no after Regional Office confirmation.
f) Online reports to Institutes, Regional Office to Admin

Online Filling & Confirmation of Examination Forms
a) Exam form covers Regular and Backlog ('X') Candidates Exam form.
b) Online generation of Seat Number for Examination and Online display of Hall ticket.
c) Online display of seating chart for Institutes.
d) Online generation and Display of Attendance sheet for Exam Centers.
e) Online generation of Timetable for Semester Exams and display of Timetable.
f) Online Module for correction of Name and Correction of Candidates Photo.
g) Online Eligibility checking and Eligibility Application and Approval.
h) Online Exemption module.
i) Online generation of RAC (Regional Assessment centers) for Theory Paper Checking.

Online Filling and Confirmation of Mark Sheets
a) Enter online marks of students in an efficient manner and reducing the errors and time required in manual process.

b) Efficiently evaluate the Candidate’s marks thoroughly through a fully automated system that not only saves lot of time but also gives faster results.
c) Responses by the examiners while filling marks checked automatically and instantly, hence no mistakes while entering marks.
d) Can generate various reports for evaluation purpose instantly when and where required.
e) E-mark sheet is designed for the institutes and RACs to enter the marks of the candidates for all types of exams conducted (Theory Marks, Theory Test Marks, Practical Marks, Practical Test Marks, Oral Marks, Term Work Marks, and Sessional Marks).
f) Design to facilitate Principal, HOD and Examiner logins in case of entry of marks for Non-Theory Exams.
g) Design to facilitate Officer in Charge, Quality Officer in Charge and Examiner logins in case of entry of marks for Theory Exams.
h) User friendly Interface.
i) Completely secure and easy way to enter marks without any mistakes.

Online Result Processing and Result Display
a) The input to Result Processing System is the data of E-Mark sheet that is entered by Examiner.
b) These marks are processed to get the final marks of each candidate seat number wise.
c) The percentage marks, Passing Class, condonation, etc is calculated.
d) This result is displayed in Result Display.

Online filling and confirmation of Verification
a) Ordinary Verification.
b) Photocopy Verification.
c) Reassessment.

4) Kuali Foundation: The Kuali Foundation is a non-profit, 501(c)(3) corporation that coordinates the development of free/open source administrative software under the Educational Community License. The name "Kuali" comes from the Indonesian word for wok. The Foundation is incorporated in the United States. Its members are colleges, universities, commercial firms, and interested organizations that share a common vision of open, modular, and distributed software systems.
Kuali initiatives

The goal of Kuali is to provide quality open source software, built for higher education, by higher education. The Foundation is working on the following initiatives:

a) **Kuali Financial Systems (KFS)**. The Kuali Financial System (KFS) project is working to create and enhance a comprehensive suite of financial software that meets the needs of all Carnegie Class institutions.

b) **Kuali Coeus (KC)**, a research administration system for higher education.

c) **Kuali Rice (Rice)** (Software Development Simplified), a suite of middleware programs (workflow, messaging, identity management), interfaces and Web services around a service bus. With the Rice components, developers can more easily build and link applications as collections of modular, interconnected services.

d) **Kuali Student (KS)**, Kuali Student (KS) is a community-source, next-generation information system designed to meet the business needs of students, faculty and institutions throughout the academic lifecycle. The software is being incrementally produced by a dedicated community of international higher education partners and distributed through the Open Source license. It achieves a rich user experience through user-centric design and offers a modular, scalable system through service-oriented architecture. Because of this modular nature, it can be integrated with legacy and enterprise systems to meet your institution’s current business needs. With its service oriented architecture, Kuali Student can be configured for how you do business today and how you’ll want to do business tomorrow.

e) **Kuali OLE (OLE)**. The Kuali Open Library Environment (OLE) (pronounced oh-LAY) is an extensible, service-driven, library management system. Currently Kuali OLE is utilizing components of KFS and Kuali Rice to build an enterprise ready java-based library management system.

f) **Kuali Ready (Ready)**, an above-campus solution for business continuity

g) **Kuali People Management for the Enterprise (KPME)**. Kuali People Management for the Enterprise (KPME) is an open-source, comprehensive HR/Payroll System built by higher education for higher education. Composed of both stand-alone and integrated modules, KPME includes Payroll, Time and Attendance, Leave Management, HR Core Transactions, Benefits Administration and Position Management.

h) **Kuali Mobility**. Kuali Mobile for the Enterprise (KME) connects the portfolio of diverse campus systems to the ever changing world of consumer devices. It uses native apps on iPhone, Android, etc. via the app stores and HTML5 to achieve 'write once, run anywhere' for low cost mobile services. KME in practice includes the following advantages:

a) University-Wide - Provides services for all members of your institution's community: Class tools for students, curriculum management features for faculty, and financial and employee services for staff.

b) Deep Integration - KME provides straightforward (not watered-down) interfaces to critical systems and data. It uses simple interfaces to tackle tasks at a high level, or drill down for more details and options.

c) Decoupled from Backend Services - Use KME to access data directly via database connections, or connect to any system that provides a compatible web service (or adaptable with KME's built-in data transformers).

**1.2 Progress**

In October 2006, the Foundation announced the first release of the open source Kuali Financial System (KFS). KFS was funded in part by a USD 2.5 million grant from the Andrew W. Mellon Foundation. It also announced that its next development project would be Kuali Research Administration. In November 2006, the rSmart Group announced that it was selling a pre-configured instance of the Kuali Financial System. The first implementation of this software, by Strathmore University in Kenya, was announced in July
The second release of Kuali Financial was in November 2007. Release 3.0 was due out in December 2008 with modules for accounts receivable and capital assets, and a large number of enhancements. In mid-2009, Colorado State University and San Joaquin Delta College went live with the first large-scale installations of full Kuali financial systems. In early 2010, Florida State University announced that it would pull out of the project because of budget cuts. Spring 2010 brought the public release of KC 2.0, and the first release of KS, made available only to the Kuali community. In addition, Rice 1.0.3 was released in the Spring of 2010, and Kuali Foundation membership grew to over 50 members. In June 2010, Indiana University announced that it was joining the KS project as MIT was stepping down, although MIT remains a member of the Kuali Foundation and a strong partner in the Kuali Coeus project for research-administration. As of late 2012, nearly one hundred institutions were running Kuali software in production.

1.3 Organization

The project was formally announced in August 2004; its initial members were Indiana University, the University of Hawaii, the National Association of College and University Business Officers (NACUBO), and rSmart. Jennifer Foutty of Indiana University is the foundation's Executive Director. Those now participating in the Kuali Foundation include more than 20 institutions of high education, and the following commercial affiliates:

a) eThority
b) HTC Global Services
c) IBM
d) The Meher Group
e) Navigator Management Partners
f) OpenCollab
g) Polus Solutions
h) rSmart
i) VivanTech, Inc.

5) Fedena:

Fedena is an open source school management software developed on Ruby on Rails framework. It is a web 2.0 web application being developed by Foradian Technologies. Fedena is used by the Education Department of Government of Kerala to automate the system and process of over 15,000 schools in the state and is named as Sampoorna. Fedena is now used in more than 40,000 institutions around the world. For the development and support of Fedena, Foradian won the MIT TR35 2012 India award for innovation in education domain.

Core Modules

The basic system of Fedena contains modules related to Courses and Batches, User Management, Human Resources, School Calendar, Student Attendance, Finance, Timetable, Student Information, Examination, Event Management, Multiple Dashboards, Employee Login, Student Admission, Teacher Login, News Management, Student/Parent Login. The system with all the core modules are available as free and opensource. Add on Modules

Various add-on modules are available as plugins to enhance the usability of the system. Presently available plugins include Hostel/Dormitory, Video Conference, Online Examination, Moodle Integration, Library, Task, Transportation, Discussion, Poll, Instant Fee, Assignment, Data Management, Placement, Custom Report, Photo Gallery, Inventory and Registration. Foradian is planning to launch an online marketplace for Fedena plugins. Integration with other services

Fedena is currently integrated with the opensource learning management system Moodle and opensource video conferencing solution Big Blue Button. Multischool version

Fedena supports multitenant architecture and is used to manage group of institutions.

Background

Origins

The Fedena project is owned by Foradian Technologies, an internet engineering company based in India.

Name and Logo

Fedena was initially named as 'foredu', as it was the first educational software from foradian technologies. But the domain name was already taken and the development team thought of something unique and web 2.0 and they selected 'fedena'. It means the coolest thing alive, usually meaning something come from royalty or from wealth. The name fedena is also motivated from Athena.
the Greek goddess of wisdom. Athena appears attended by an owl. So the creative team of fedena selected owl as the symbol and derived the logo from the face of an owl. Owl is also the symbol of wisdom and knowledge in many mythologies and beliefs.

7) iON Education Solutions

"iON has established strong credentials for fool-proof delivery of ICT Services. We chose iON Education Solution consisting of 30 modules covering university processes including admission, fee collection, teaching learning processes, finance and accounts, purchase and inventory, human resource management system, payroll management, transport, hostel, library and other allied activities. iON is based upon cloud computing and is a fully secured application and hence, having high reliability. Implementation of iON has enriched the reach of data and subsequent analysis efficiently across all modules."

When an institution looks at its campus as a site of heritage, it actually cherishes its way of teaching. With Google and Wikipedia helping students, the old ways of teaching are often put to question. There are two sides to this - Teaching traditions define a school; new teaching channels define its influence.

8) CollegeExcel:
Glodyne Technologies is a Mumbai based Managed Application Services provider company primarily operating in the domestic education space. Broadlyline’s has two Web enabled software product namely School Excel and College Excel.

College Excel is truly comprehensive e-governance software that covers the entire gamut of campus activities in a College. The software is aimed at bringing in process refinement, efficiency, accountability and control in a College. The extensively researched and well-tested architecture of College Excel ably suits the needs of variety of institutions of higher education ranging from the traditional degree colleges (Arts, Science and Commerce Colleges) to the colleges running technical and professional courses.

9) IEMS

Integrated Education Management System (IEMS) is comprehensive software system that covers all administration and operations of the educational institution. IEMS is a one package system that eliminates the need for multiple software system. Its a combination of Student information system(SIS), ERP, CRM, LMS, Academics Management System.

Revolution Next Technologies Pvt. Ltd was started in 2008 by Rahul Sharma, Director and CEO of the company. We are product based Software Company and focus on internet based products. We are one of the leading start up companies of North India which focuses on Hi-Tech Software Development. We are involved in High-end custom Software Development services for Domestic & Overseas Clients. Some of our products and solutions are Integrated Education Management System, E-Learning System, ERP, CRM, Web Solutions, Custom Portal Development, Content Management & E-Learning Solutions, Financial Products, Business Intelligence, Artificial Intelligence products. Our highly skilled team of software engineers in India delivers high quality software products and services to consumers and companies.

10) Smart Campus 4.0

ThinkNEXT Technologies Private Limited (An ISO 9001:2008 Certified Company), is emerging as most innovative company in Education Domain in India. The Management of ThinkNEXT Technologies Private Limited has wide experience more than nine years in education domain. Over the years, we have worked very closely with Universities, Group of Colleges and other Institutions. We have wide experience working with eminent Educationists, Managements, Directors, Principals, Head of Departments, other Staff Members, Parents and students. Therefore we do not sell only software Modules but an innovative system which has more importance than just ERP software modules. Today Smart Campus solutions are a need of hour for every University/Group of Colleges or an Institution to make edge over others and maintain a lead over their competitors. Our Research and Development team is committed to make your institute(s) to maintain lead over their competitors.

11) eCampus

ECampus (Comprehensive Education Institute Management System) is a completely integrated 100% web based EDUCATION ERP for Institutions/Organizations. eCampus is a synonymous with education institute administrative solution and ERP information system.

Why eCampus

- For Management
For Parents

For Faculty

For Students

Our solution helps schools, institutes, Colleges and Universities to manage entire student life cycle – Admissions, Registrations, Student records, financial aid and management, course delivery, development, placement and Alumni Relations. eCampus also includes a complete suite of portals to provide students, faculty and alumni with 24/7 access to information and services like Online Application, Online Registration, Online Fee Payment and Register for Online Examination.

REFERENCES

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[14] Kuali Foundation releases open source financial system, launches a research administration system project, announces new board members and corporate partners", press
Abstract: The objectives are:
- Providing checklist for Inventory Setup
- Deeping understanding and creating body of knowledge on Items in LS Retail NAV
- Placing the Item in the LS Retail NAV hierarchy
- Showing the attributes and other features for Item

I Introduction

Items are the fundamental unit in the LS Retail System. LS Retail includes several features concerning item sales at the point of sale. Following is a list of the main features:

1. Item Divisions

Item Divisions (called Divisions in LS Retail) represent the most general grouping of items. They are used for ‘Open to Buy’ and some reports.

2. Item Categories

Item Categories represent the next level for grouping of items, making it possible to examine sales statistics on a broad perspective. You can also use Item Categories when setting up Open Department sales. Item Categories provide default posting setup groups for the related items.

3. Product Groups

Product Groups provide basic parameter that the system assigns to the item when you select a Product Group for it. Product Groups can control the barcode construction and variant groups for the retail items included in each group. They can also make the process of locating items in sections and shelves easier and control the location distribution for the items. Furthermore, discount offers can be created at Product Group level, as well as at variants and retail item levels.

4. Variants

Retail items can have 6 different variant dimensions like color, size and style. Each combination can be represented by a unique barcode. Thus, statistical results of sales by variants are available.

5. Barcodes

Retail items can be represented with one or more barcodes in addition to the item number itself. Multiple barcodes are essential, that is, if the same product comes from different manufacturers, or if the item has different variants. The system can generate barcodes for all variant combinations with a mask that takes barcode numbers from a number series. The system also supports EAN 8, EAN 13, UPC-A and UPC-E standard barcodes.

6. Printing

The system supports the generation of Item Labels and Shelf Labels, using pre-formatted reports.

II Checklist for Inventory Setup

When you set up inventory in LS Retail, you must enter certain information before you can start running the system. Certain setup is mandatory; other is optional.

For setting up inventory in LS Retail follow the tasks according to the order given, note that it has been assumed that posting groups have already been set up in the General Ledger of Microsoft Dynamics NAV, as well as tasks carried out in the Retail Setup, therefore those tasks are not mentioned in the checklist [1].

Table 1: Checklist for Inventory Setup

<table>
<thead>
<tr>
<th>S no</th>
<th>Task/Overview</th>
<th>Mandatory</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Setting up Units of Measure</td>
<td>Yes</td>
</tr>
<tr>
<td>2</td>
<td>Setting Up Item Division</td>
<td>No</td>
</tr>
<tr>
<td>3</td>
<td>Setting Up Item Categories</td>
<td>Yes</td>
</tr>
<tr>
<td>4</td>
<td>Setting Up Product Groups</td>
<td>Yes</td>
</tr>
<tr>
<td>5</td>
<td>Setting Up Price Groups</td>
<td>Yes</td>
</tr>
<tr>
<td>6</td>
<td>Setting Up Special Groups</td>
<td>No</td>
</tr>
<tr>
<td>7</td>
<td>Setting Up Item Attributes</td>
<td>No</td>
</tr>
<tr>
<td>8</td>
<td>Setting Up Seasons</td>
<td>No</td>
</tr>
<tr>
<td></td>
<td>Setting Up Events</td>
<td>No</td>
</tr>
<tr>
<td>---</td>
<td>------------------</td>
<td>----</td>
</tr>
<tr>
<td>10</td>
<td>Creating Retail Items</td>
<td>Yes</td>
</tr>
<tr>
<td>11</td>
<td>Registering Item Prices</td>
<td>Yes</td>
</tr>
<tr>
<td>12</td>
<td>Linking Items</td>
<td>No</td>
</tr>
<tr>
<td>13</td>
<td>Setting Up Variant Framework</td>
<td>No</td>
</tr>
<tr>
<td>14</td>
<td>Creating Variants</td>
<td>No</td>
</tr>
<tr>
<td>15</td>
<td>Setting Up Barcode Mask Characters</td>
<td>No</td>
</tr>
<tr>
<td>16</td>
<td>Setting Up Barcode Masks</td>
<td>No</td>
</tr>
<tr>
<td>17</td>
<td>Entering Barcodes</td>
<td>No</td>
</tr>
<tr>
<td>18</td>
<td>Assigning Sections and Shelves to Items</td>
<td>No</td>
</tr>
<tr>
<td>19</td>
<td>Setting Up Item Distribution</td>
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</tr>
<tr>
<td>20</td>
<td>Specifying Receipt Texts for Items</td>
<td>No</td>
</tr>
<tr>
<td>21</td>
<td>Setting Up Item Labels</td>
<td>No</td>
</tr>
<tr>
<td>22</td>
<td>Setting Up Shelf Labels</td>
<td>No</td>
</tr>
<tr>
<td>23</td>
<td>Assigning Extra Print Setup to Retail Items</td>
<td>No</td>
</tr>
</tbody>
</table>

1. Item Divisions

The use of Item Divisions is not mandatory. It can be used for Open-To-Buy, reports, and retail item search.

a. To Set Up Item Divisions:
1. Click LS Retail – Back Office, Setup, Item, Groups, and Divisions.
2. Press F3 to enter a new Item Division.
3. Fill in the Code and Description fields.

Fig 1: To Set Up Item Divisions

2. Item Categories

There are two more levels of Retail Product Grouping in LS Retail:

- **Item Categories**: The higher level. Each Item Category contains a number of Product Groups.
- **Product Groups**: The lower level of grouping. Each Product Group contains a number of retail items. It is important to have a well-defined Item Category structure, before you start to set up Item Categories. Before setting up Item Categories you should define how detailed the Item Category structure should be in order to serve best the needs of your business. Then you Create Item Categories and assign closely related Product Groups to the same Item Category.

To ensure the correct handling of retail items, you need to set up Item Categories. You can use Item Categories to:

- Group Product Groups
- Define default Posting Groups
- Base POS cost calculation on Item Categories
- Collect and view statistics on Item Category level
- Set up an Open Item category sale

b. To Set Up Item Categories:
1. Click LS Retail – Back Office, Setup, Item, Groups, and Item Categories.
2. Press F3 to enter a new Item Category.
3. Fill in the Code and Description fields.
4. Fill in the Default Profit % field if needed. To base POS cost calculation on an Item Category you must select the Item Category Based option in the POS Cost Calculation field for each retail item involved.

Fig 2: To Set Up Item Categories
Repeating steps 2 to 4 for each Item Category you want to set up.

Microsoft Business Solution NAV does not allow a sale on open item groups. Only sales on items are possible. Nevertheless you can set up open item category sale or open product group sale, meaning that when you have an item that does not have an item number registered, you can sell the item as a part of the Item Category or a Product Group.

**Note:**

- When selling an open item category sale, statistics will not be retained on item level, but on Item Category level, also; the inventory will not be updated.

### b. To Set Up Open Item Category Sale:

Create a retail item. In the **Description** field, insert a description of the Open Item Category sale.

1. In the **Item Category Code** field on the Retail Item Card, select the relevant Item Category.
2. On the **POS tab** in the Retail Item Card window, select the option **Must Key in New Price** in the **Keying in Price** field.
3. On the **Invoicing** tab, select the required **POS Cost Calculation** and **No Stock Posting**.

When performing an Open Item Category sale, you select the item created above as the Item Category to be sold. You can make a similar setup for a Product Group.

### 3. Product Groups

Product Groups provide a convenient way of handling barcode generation and checking, as well as supporting default variants for the retail items included in the group. It is important to have a well defined Product Group structure, before you start to set up Product Groups. Before setting up Product Groups you should define how detailed the Product Group structure should be in order to serve best the needs of your business.

Then you create Product Groups and assign closely related retail items to the same Product Group. This is especially true for retail items that have variants.

#### a. To Set Up Product Groups:

1. Click **LS Retail – Back Office, Setup, Item, Groups, and Product Groups**.
2. Press **F3** to enter a new Product Group.
3. Fill in the **Code** and **Description** fields.
4. In the **Item Category** field, click the Assist Button to see the **Item Categories** window. Select the relevant Item Category, and then click **OK** to copy it to the field.
5. Fill in the remaining fields as appropriate.

Repeat steps 2 to 5 for each Product Group you want to create.

When you select a Product Group for a retail item, the Item Category, the posting groups, the Section and shelf location, the barcode mask and variant groups of the Product Group will all be automatically assigned to the item. When you make changes to these features after you have assigned items to the group, you can use a function to copy these features specifically to the items.

#### b. To Copy from Product Groups to Retail Items:

2. Browse to the relevant Product Group and click **Functions, Copy from Product Group to Items**.
3. On the **Options** tab, place a check mark in the fields for which settings you want to copy from the Product Group to its items.
4. Click **OK** to run the function. The program will copy the requested information to the retail items in the group.

**Note:**

- Before copying information from Product Groups to retail items, the relevant retail items must already have been assigned to the Product Groups.

### 4. Retail Items

Basically, a retail item is created and set up as an ordinary item in the inventory system. For example you can use the standard Navision Bill of Material item setup and explode a retail item’s BOM when posting statements.

When you have entered the necessary information and closed the window, the system creates an action in the Actions table. The new item will then be on file and ready to be sold, after the system next exports retail item date to the POS terminals.

#### a. To Create Retail Items:

1. Click **LS Retail – Back Office, Retail Item Card.**
2. Press **F3** to enter a new retail item.
3. Fill in the **No.** field, by selecting the appropriate number series.
4. Fill in the **Description** field.
5. If needed, fill in the **Base Unit of Measure** field, by selecting the appropriate unit of measure.
6. In the **Product Group Code** field, select the relevant Product Group code. Notice that the Item Category field is automatically filled with the Item Category of the Product Group.
7. On the **Invoicing** tab, note that the **Gen. Prod. Posting Group, VAT Prod. Posting Group** and **Inventory Posting Group** fields have all been copied from the Product Group as default values. You can change these groups for each retail item.
8. Fill in the **Cost** and the **Unit Price Including VAT**.
9. If the item needs to be weighed on a scale at sales time, place a check mark in the **Scale Item** field on the **POS** tab.
10. If you like to work with additional prices, click **Pricing** and select the relevant sales code.
11. Fill in other fields in the **Item Card** window as needed. 
Repeat steps 2 to 11 for each retail item you want to create.

5. **Item Linking**

You can link retail items together, so that every time the main retail item is sold, the retail items that are linked to it are sold also. This is for example used when selling drinks in bottles, then the bottles are linked to the drinks so that every time the drink is sold, the bottle is sold with it.

Linking can for example be used when selling a bed. Then you can link the mattress, frame and legs to the bed item, so that each time the bed is sold (bearing zero price), the other items are sold automatically with it.

a. **To Link Items:**

1. Click **LS Retail – Back Office, Retail Item Card**. The **Retail Item Card** window appears.
2. Click **Item, Item Linking, Linked Items**. The **Linked Items** window appears.
3. In the **Linked Item No.** field, select the relevant item from the **Item List** window.
4. In the **Unit of Measure** field, select the Unit of Measure you like to have the item linking for.
5. In the **No. of Items** field, fill in how many units of this item should be linked to the main item; that is the item in the **Linked Item No.** field.
6. If the linked item shall be returned later – for example an empty bottle with bottle deposit paid for – then the **No. of Items should be negative and on the item (the bottle item) the parameter Qty. Becomes Negative should be set**. Then when selling the main item and the linked item, the linked item is sold “positive”. When returning the linked item, it is negative.

**Note:**

It is important to make a distinction between the main item and the linked item. Each time the main item is sold, the item that is linked to it, the linked item, is sold also. The reverse does not apply, that is, when the linked item is sold, the main item is not sold also. The main item cannot be linked to other items, that is, it can never appear as a linked item. Neither can you link an item to itself. When linked items are sold, the retail items that are linked to it are sold also. You can view which retail items have been linked to other retail items.

b. **To View Linked Items:**

1. Click **LS Retail – Back Office, Retail Item Card**. The **Retail Item Card** window appears.
2. Click **Item, Item Linking, Where-Linked List**.
3. In the **Where-Linked List** window, you can see the main item to which item the relevant item is linked. The **No. of Items** field indicates how many units of the item are linked to the main item.

6. **Special Groups**

A retail item can be a part of special groups. They are used for extra grouping out of the static item hierarchy.

a. **To Set Up Special Groups:**

1. Click **LS Retail – Back Office, Item, Groups, And Special Groups**.
2. Press F3 to enter a new Special Product Group.
3. Fill in the **Code** and **Description**.

b. **To Assign Items to Special Groups:**

1. Click **LS Retail – Back Office, Retail Item Card**. The **Retail Item Card** window appears.
2. Click **Item, Special Groups**.
3. In the field **Special Group** select the special group. Repeat steps 3 to 5 for each special group you would like to assign to the item.

Special Groups can be used in LS Retail Standard for the Retail Item Search, definition of station printing and as a filter on the POS Command DYNMENU. You can link a special group to an Item Hierarchy. This gives the option to see the special groups in the Sales History. Optionally you can decide in the **Back Office, Setup, Retail Setup, Sales History** Tab to ‘Include Special Groups’. In that case you do not need to link it to a Item Hierarchy but still can see them in the Sales History. Special Groups can be assigned to Seasons and Events.

7. **Item Attributes**

Item Attributes are used to assign any number of additional fields for items:

- Text, Numeric, Amount & Dates
- Lookups to Navision tables
- Lookups with pre-defined options
- Define valid input
- Define number of instances allowed per item, or unlimited per. Item

They are used for filtering and lookups without extra development.

Item Attributes are automatically added to items based on:

- Item Category
- Item Product Group

a. **To Set Up Item Attributes:**

1. Click **LS Retail – Back Office, Setup, Item, Attributes, Attributes Setup**.
2. Press F3 to enter a new Attribute.
3. Fill in the **Code**.
4. Set the other parameter as needed.

b. **To Assign Item Attributes to Item Categories and Product Groups:**
1. Click LS Retail – Back Office, Setup, Item, Attributes, Item Attribute Settings.
2. Select the Attribute Code and the Item Category and Product Group you would like to assign it to. Item Attributes are set automatically when item created based on item category or based on product group. Item Attributes can be created manually as well. The Item Attributes must be linked to item category or product group before.

c. To Assign Item Attributes to an Item:

1. Click LS Retail – Back Office, Retail Item Card. The Retail Item Card window appears
2. Click Item, Attributes.
3. To create an Attribute press F3.
Item Attributes are more flexible than Special Groups when it comes to specific configurations. They can be used in LS Retail Standard for the Retail Item Search.

8. Seasons

A Season is used to group items together for merchandising purposes. A Season has a Starting Date and an Ending Date. A Special Group can be assigned to a season. The Season is used in the Retail Item Search and in a sales report by Season. You can populate promotions and discount offers with items from a season via a season.

a. To Set Up Seasons:

1. Click LS Retail – Back Office, Setup, Item, Groups, Seasons.
2. Press F3 to enter a new Season.
3. Fill in the Code and Description.
4. Select a Starting and Ending Date.
5. If required, select a special group for this Season.

b. To Assign Items to Special Groups:

1. Click LS Retail – Back Office, Retail Item Card. The Retail Item Card window appears
2. On the Merchandising tab select the Season Code.

9. Events

An Event could be for example an End-of-Season sale or a promotion of items for Easter. The Event has a starting date and an ending date. A Special Group can be linked to an Event. The promotion and discount offer allow importing items which are assigned to an Event and a report gives sales information for an Event. A Special Group can be linked to an Event.

a. To Set Up Events:

1. Click LS Retail – Back Office, Setup, Item, Groups, Events.
2. Press F3 to enter a new Event.
3. Fill in the Code and Description.
4. Select a Starting and Ending Date.
5. If required, select a special group for this Event.

b. To Assign Events to an Item:

1. Click LS Retail – Back Office, Retail Item Card. The Retail Item Card window appears
2. Click Item, Events.
3. In the field Event Code, select the Event.
4. Fill in a Comment if required.

10. Item Families

Item Families are a basic item grouping mechanism to give the retailer a possibility for additional item grouping [2].

a. To Set Up Item Families:

2. Press F3 to enter a new Item Family.
3. Fill in the Code and Description.

b. To Assign Item Families to an Item:

1. Click LS Retail – Back Office, Retail Item Card. The Retail Item Card window appears
2. On the General tab you can select the Item Family Code.

III Variant Framework

Variants allow you to handle retail items that have up to 6 different variant dimensions. By assigning a unique barcode for each variant combination, you can scan the barcode at the POS terminal and let the program find which variant of the item is being sold. You can therefore collect and view statistics of variant sales.

When you have assigned barcode masks to Product Groups or items the program uses the barcode mask to generate barcodes automatically for each variant combination. This can be extremely useful if there are many variant dimensions possible as the combination increase greatly with added variant codes. Otherwise, the combinations will have to be manually assigned to each barcode representing a variant.

When you have assigned a variant framework code to an Product Group you can create variants for the retail items in the Product Group. You can also assign a variant framework code to retail items and then create variants for retail items.

1. Variant Framework Codes

Before you set up variant dimensions, variant framework codes must have been set up.
To Set Up Variant Framework Codes:

2. Insert a Framework Code and fill in the description.
4. Fill in the Barcode Mask field in case you like to use the barcode mask.
5. Fill the other fields as needed

Repeat steps 2 to 5 for each Variant Framework Code you want to set up.

In the Variant Framework Combinations you define the combinations of Variant Dimension Base Values you like to use for the specific Variant Framework Codes.

To Set Up the Variant Framework Combinations:

2. Click Settings, Combinations.
3. In the Code field select the Dimension Setups you like to have included in the combination.
4. In the sub-form you can see the Values, for example colors.

Variant Framework Base Values

Base values for the variant framework can be values for size, color, style and others.

To Set Up the Variant Framework Base Values:

2. In the Code field define the Base Value Code like COLOR, SIZE and STYLE.
3. In the Sub Form specify the Values like different colors, sizes and styles.
4. For colors you can see the color by clicking the AssistEdit button on the Color field.
5. The logical order allows resorting of the values.

Assigning the Variant Framework

Before creating variants for retail items you must either assign a variant framework code to the Product Group the retail item belongs to, or assign the variant group combination to the item itself. If you want to create item variants for all items in a Product Group you should assign variants to the Product Group. When assigning the Product Group to the retail item, the item will inherit the variant group combination, which you can change if needed.

To Assign a Variant Framework Code to Product Groups:

2. Browse to the Product Group you want to assign a variant framework code for.
3. Click Product Groups, Card.
4. In the field Variant Framework Code select the required Variant Framework Code.

To Assign a Variant Framework Code to Retail Items:

1. Click LS Retail – Back Office, Retail Item Card. The Retail Item Card window appears.
2. Browse to the retail item you want to assign a variant framework code to.

Creating Variants

A Variant combination is created for each item. Before creating variants you must either assign a Variant Framework Code to the Product Group the retail item belongs to, or assign the variant group combination to the item itself.

To Create Variants:

1. When you assign the Variant Framework Code to a retail item, the system will guide you through the selection.
2. You can select the different Variant Dimension Values coming from the Variant Framework code.
3. The system will create Variants according to the selection and the Suffix / Sequence No. from the Retail Setup.
4. Optional you can follow this way:
5. Click LS Retail – Back Office, Retail Item Card.
The Retail Item Card window appears.
6. Browse to the retail item you want to assign a variant framework code to.
7. Click Item, Variant Framework, Functions, Register Defaults.

5. Units of Measure

Before you start creating retail items you should set up units of measure codes, which you assign to retail items. You can set up an unlimited number of units of measure codes.

a. To Set Up Units of Measure:

1. Click LS Retail – Back Office, Setup, Item, Unit of Measure, Units of Measure. The Units of Measure window appears.
2. Fill in the Code and Description fields.
3. If you want the unit of measure to be the one the scale and barcodes use as a base unit of measure, place a checkmark in the Weight Unit of Measure field.

To calculate the item price for a comparison unit, you need to set up comparison units of measure and the conversion factors between them. For example, to calculate kilogram price for an item that is sold in a 200g package, you need to set up the conversion factor between the unit of measure of the item and the comparison unit. When you have set up comparison units of measure, you can assign those to items and have the program calculate the item price for the comparison unit, for example, the kilogram price for an item that is sold in a 200g package. Setting up comparison units of measure makes it possible to compare prices of items even if they are of varying quantities and units of measure.

Once you have defined the conversion factor between one unit and other two units, that is, between A and B, and A and C, the program will automatically calculate the conversion factor between B and C. This saves you a considerable amount of work when setting up comparison units of measure.

b. To Set Up Comparison Units of Measure:

1. Click LS Retail – Back Office, Setup, Item, Unit of Measure, Comparison Units of Measure. The Comparison Units of Measure window appears.
2. Press F3 to enter a new comparison unit of measure.
3. Fill in the Code and Description fields.
4. Click Comp. Unit, Conversion, for a unit you want to use as bases for conversion. The Conversion window appears. Note that the program has automatically entered the conversion factor 1 between the chosen base comparison unit and itself.
5. In the Comparison Unit Code field, select a unit of measure.
6. Fill in the conversion factor in the Conversion Factor field, keeping in mind the equation: Base Unit = Comparison Unit * Conversion Factor.
Repeat steps 5 and 6 for additional units you want to convert to the base unit.

Once you have completed entering conversion factors between the base unit and the selected units, the program has created conversion entries and calculated conversion factors between each pair of the selected units.

Example:
You have set up three comparison units, Liter, Milliliter and Centiliter. You first select Liter to set up its conversion values. The conversion factor between Milliliter and Liter is 1000 and the conversion factor between Centiliter and Liter is 100. When you have entered this information in the Conversion window for Liter, the program has already created a conversion value entry for Milliliter and Centiliter. If you later add another unit, like fluid ounces, you only need to set up a conversion factor with one unit, such as Liter. The program then calculates and creates entries for the other units connected with Liter. If the base unit is kilogram and the comparison unit is gram, the conversion factor is 1000. If the base unit is gram and the comparison unit is kilogram, the conversion factor is 0.001.

6. Comparison Prices

You can have the system calculate a comparison price for an item, based on a particular comparison unit of measure. This enables you to compare the prices of items that are sold in units of different size.

a. To Calculate Comparison Prices:

1. Click LS Retail – Back Office, Retail Item Card, the Retail Item Card window appears.
2. Browse to the item, for which you want to calculate comparison prices.
3. Click the Comp. Price tab.
4. Fill in the Base Comp. Unit Code field, by selecting the relevant comparison unit code.
5. In the Qty. Per Base Comp. field, enter the relevant quantity.
6. Fill in the Comparison Unit Code field, by selecting the comparison unit code you want to compare the base comparison unit to.

The program automatically calculates the comparison price. From now on, when you change the unit price of the item, the program will update the comparison price accordingly. Please note that this is only valid for the item price on item card level which may not be the active sales price.

Example:
A comparison price in kilograms for an item sold in 200 g units, at the price of 20, is 100. The kilogram price can then be compared with the kilogram price for other items. When you have calculated the comparison price for a retail item, you can print it on shelf labels.

7. Competitors

You can set up competitors and register the competitor price for a specific retail item.

a. To Set Up Competitors:

1. Click LS Retail – Back Office, Periodic Activities, Retail Competitors. The Retail Competitors window appears.
2. Fill in the Code and Description fields.
3. Fill in other fields as needed.
You can keep track of your competitors’ prices of each retail item. You register the prices either from each retail item or from a competitor. You must register a competitor before registering competitor prices.

b. To Register Competitor Prices:

1. Click LS Retail – Back Office, Periodic Activities, Retail Competitors. The Retail Competitors window appears.
2. Browse to the relevant competitor.
3. With a DrillDown in the Date of Last Price Check field you can open the Competitors Ledger Entry list.
4. Fill in the Item No. field by selecting an item number.
5. Fill in the Date and Price fields.

IV Sections and Shelves

To have access to sales information for sections and shelves, you need to assign sections and shelves to retail items or Product Groups. You can assign as many sections and shelves to an item or Product Group as you want. However, you can only get sales information for one section and shelf per item, unless you have a way of knowing at the POS terminals from which section and shelf the item was taken. If this is the case, you need to adapt this method to the LS Retail system.

a. To Assign Sections and Shelves to Product Groups:

2. Click Product Groups, Card.
3. Browse to the relevant Product Group and click Prod. Group, Section Location. The Product Group Section Locations window appears.
5. For one, and only one of the section locations, place a check mark in the Shows Statistics field. The program will keep information of sales for the relevant Product Group for this section and shelf assignment only. Repeat step 4 for each section and shelf you want to assign to the Product Group.

Note:-

When you assign a Product Group to a Retail Item, the item will automatically have the same section and shelf location as the Product Group. You can adapt this assignment for each item.

b. To Assign Sections and Shelves to Items:

1. Click LS Retail – Back Office, Retail Item Card. The Retail Item Card window appears.
2. Click Item, Store Information, Section Locations. The Item Section Locations window appears.
3. If the Product Group the selected item belongs to has been assigned section locations the program has already copied the section location of the Product Group to the item. In that case, you can change the Section Code and Shelf Code fields as needed.
4. If the Product Group has not assigned section location, fill in the Section Code and Shelf Code fields.
5. For one, and only one of the section locations, place a check mark in the Shows Statistics field. The program will keep information of sales for the relevant item for this section and shelf assignment only. Repeat step 4 and 5 for each section and shelf you want to assign to the Product Group.

1. Retail Item Printing

Retail Item Printing is functionality for label printing and extra printing. Labels are generated by using actions (or preactions if used for replication). The system scans the Action table for any data changes that require new labels to be printed. The system contains information about all labels needed in the future. Therefore you can order labels for both the beginning date of a periodic offer and at the ending date. Label orders are specific to each store, item, variant and unit of measure. New labels to print should be created on:

- Price change
- Comp. price change
- Description change (item or variant)
- Barcode change (the one on the retail item card)
- Label report change
- New variant added
- New unit of measure is marked to be printed
- The label reports in the system are for reference. If the item price is higher than 100,000 you need to modify the report.
The Extra Print Setup allows defining extra print texts like a warranty card or item care instructions. These texts can be assigned to items for printing on the OPOS POS printer.

2. Label Printing Basic Definitions

Basic definitions for ordering and printing of labels are found at some places in LS Retail:
- Retail Setup, Labels tab
- Store Card, General tab
- Retail Item Card, General tab
- Item Unit of Measure Card

3. Item and Shelf Label Reports

In order to print labels, you need to define and specify reports. The label reports in the demo system are for reference. If the item price is higher than 100,000 you need to modify the report. For printing barcodes you need to have a barcode font. LS Retail does not provide you with any barcode fonts.

a. To Set Up Shelf Labels and Assign Reports

1. Click LS Retail – BackOffice, Setup, Labels, Shelf Label Reports
2. Choose the label reports and assign a Label Code to them. This code is the reference used in the system.
3. Specify the Report ID to define which report shall be used in order to print the label.

b. To Set Up Item Labels and Assign Reports

1. Click LS Retail – BackOffice, Setup, Labels, Item Label Reports
2. Choose the label reports and assign a Label Code to them. This code is the reference used in the system.
3. Specify the Report ID to define which report should be used in order to print the label.

4. Shelf Labels

Before printing shelf labels, you must assign shelf labels to retail items. When you assign shelf labels to items, you can set up which shelf label report you want to use to print shelf labels for individual items.

a. To Assign Shelf Labels to Items:

1. Click LS Retail – Back Office, Retail Item Card, the Retail Item Card window appears.
2. Browse to the item you want to assign a shelf label report to.
3. Click Item, Text and Printing Setup, Shelf Label Setup.
4. Fill in the Store Group field by selecting the relevant store group.
5. Fill in the Label Code field.
6. If you like to have a different label type for printing promotion labels, then you can select another label code in the field Red Flag Label Code.
7. Fill in other fields as needed.

b. To Print Shelf Labels:

1. Click LS Retail – Back Office, Labels, Shelf Label Print. The Shelf Label Printing window appears.
2. Select a function to order labels.
3. Select the label code and print the label.

There are 4 functions for ordering labels:
- Create Needed Labels. Runs through the Action Table and checks if any data has changed that needs to be updated on a label.
- Create Shelf Labels by Item. Creates labels for items according to selected filter.
- Shelf Labels by Purch.doc. Creates labels from the lines by Purchase Document.
- Shelf Labels by Posted Purch.doc. Creates labels from the lines by posted Purchase Invoice.

Note:-
When opening the Shelf Label Printing window, the system will check for labels to be generated.

5. Item Labels

Before printing item labels, you must assign item labels to retail items. When you assign item labels to items, you can set up which item label report you want to use to print item labels for individual items.

a. To Assign Item Labels to Items:

1. Click LS Retail – Back Office, Retail Item Card, the Retail Item Card window appears.
2. Browse to the item you want to assign an item label report to.
3. Click Item, Text and Printing Setup, Item Label Setup.
4. Fill in the Store Group field by selecting the relevant store group.
5. Fill in the Label Code field.
6. If you like to have a different label type for printing promotion labels, then you can select another label code in the field Red Flag Label Code.
In many cases, a company's inventory represents one of its largest investments, along with its workforce and locations. Inventory management software helps companies cut expenses by minimizing the amount of unnecessary parts and products in storage. It also helps companies keep lost sales to a minimum by having enough stock on hand to meet demand.

2. Increased efficiency

Inventory management software often allows for automation of many inventory-related tasks. For example, software can automatically collect data, conduct calculations, and create records. This not only results in time savings, cost savings, but also increases business efficiency.

3. Warehouse organization

Inventory management software can help distributors, wholesalers, manufacturers and retailers optimize their warehouses. If certain products are often sold together or are more popular than others, those products can be grouped together or placed near the delivery area to speed up the process of picking.

4. Updated data

Up-to-date, real-time data on inventory conditions and levels is another advantage inventory management software gives companies. Company executives can usually access the software through a mobile device, laptop or PC to check current inventory numbers. This automatic updating of inventory records allows businesses to make informed decisions.

5. Data security

With the aid of restricted user rights, company managers can allow many employees to assist in inventory management. They can grant employees enough information access to receive products, make orders, transfer products and do other tasks without compromising company security. This can speed up the inventory management process and save managers' time.

6. Insight into trends

Tracking where products are stocked, which suppliers they come from, and the length of time they are stored is made possible with inventory management software. By analyzing such data, companies can control inventory levels and maximize the use of warehouse space. Furthermore, firms are more prepared for the demands and supplies of the market, especially during special circumstances such as a peak season on a particular month. Through the reports generated by the inventory management software, firms are also able to gather important data that may be put in a model for it to be analyzed.

The main disadvantages of inventory management software are its cost and complexity.

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6. Fill in the other fields on the Item Label Setup window as needed.

b. To Print Item Labels:

1. Click LS Retail – Back Office. Labels, Item Label Print. The Item Label Printing window appears.
2. Select a function to order labels.
3. Select the label code and print the label.

There are 4 functions to order labels:

- Create Needed Item Labels. Runs through the Action Table and checks if any data has changed that needs to be updated on a label.
- Create Item Labels by Item. Creates labels for items according to selected filter.
- Item Labels by Purch.doc. Creates labels from the lines by not posted Purchase Document.
- Item Labels by Posted Purch.doc. Creates labels from the lines by posted Purchase Invoice.

Furthermore item labels can be printed directly from:

- A Retail Purchase Order
- A Store Posted Purchase Invoice.

Note:-

When opening the Item Label Printing window, the system will check for labels to be generated.

6. Label Quick Print from the Retail Item Card

Shelf and Item Labels can be printed straight from the Item Card. In both cases the system opens ma form showing what will be printed after the setup on the item. You can modify the records, change the quantity, delete or add as you wish. To print just click Print and all labels on the form will be printed. Note that this can also be done from the Item Search card.

7. Extra Prints

You can assign extra prints of receipts for given retail items. In order to assign extra prints you must first have to set up extra prints.

a. To Assign Extra Print Setup to Retail Items:

1. Click LS Retail – Back Office, Retail Item Card. The Retail Item Card window appears.
2. Click Item, POS, Extra Print Setup. The Extra Print Setup window appears.
3. Fill in the Setup ID field, by selecting an extra printout setup from the POS Print Setup List window.

   A. V Advantages and disadvantages

There are several advantages to using inventory management software in a business setting [3].

1. Cost savings
1. Expense

Cost can be a major disadvantage of inventory management software. Many large companies use inventory management software, but small businesses can find it difficult to afford it. Barcode readers and other hardware can compound this problem by adding even more cost to companies. The advantage of allowing multiple employees to perform inventory-management tasks is tempered by the cost of additional barcode readers. Use of smart phones as QR code readers has been a way that smaller companies avoid the high expensive of custom hardware for inventory management.

2. Complexity

Inventory management software is not necessarily simple or easy to learn. A company’s management team must dedicate a certain amount of time to learning a new system, including both software and hardware, in order to put it to use. Most inventory management software includes training manuals and other information available to users. Despite its apparent complexity, inventory management software offers a degree of stability to companies. For example, if an IT employee in charge of the system leaves the company, a replacement can be comparatively inexpensive to train compared to if the company used multiple programs to store inventory data.

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Revolutionary techniques in theory of automata
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Abstract:
Theory of automata is the study of self operating virtual machines which helps to understand the input and output process of computation. It may or may not include intermediate stages. There are many computational problems in real world applications which can be designed and modelling efficiently using automata theory because of its computing power. Automata and computational theory are used in diversity of new field such as designing of games, XML, biology, neural network and decidability like P vs NP problems. This paper is the survey of various revolutionary techniques in automata. The intention of this paper is to give an overview of new techniques of automata, to give more new research ideas in this field.

Keywords: Theory of automata, XML, P vs NP, Game, Molecular.

1. Introduction
The theory of Automata has undergone a number of evolutions in a short span of time. From its beginning in the 1960s as an outgrowth of mathematical logic and information theory, it evolved into a branch of mathematics where one looks at classical problems with the aesthetics of computational complexity and asks new questions concerning non-determinism, randomness, approximation, interaction, and locality. It then took a foundational role in addressing challenges arising in computer systems and networks, such as error-free communication, cryptography, routing, and search, and is now a rising force in the sciences: exact, life, and social. [1]

What Automata Is About
Automata theory is the study of abstract machines and automata, as well as the computational problems that can be solved using them. It is a theory in theoretical computer science, under discrete mathematics (a section of Mathematics and also of Computer Science). Automata come from the Greek word αὐτόματα meaning "self-acting". Automata Theory is the study of self-operating virtual machines to help in logical understanding of input and output process, without or with intermediate stage(s) of computation (or any function/ process). As the last two questions suggest, Theory of automata has increasingly been branching out, applying its intellectual toolkit to biology, economics, physics, and many other fields. But at the same time, it has also maintained a core of fundamental ideas and problems of its own. Theory of Automata maintains a core of fundamental ideas and problems such as the famous P vs NP problem or speeding up algorithms for traditional problems in graph theory, algebra, and geometry. [1] [2]

This paper is organized in 3 sections. Background details of automata have been discussed in section 2. Section 3 is about the Revolutionary techniques in automata.

2. Background
Theory of automata is a theoretical branch of computer science. Its existence found since 20th century. It’s a study of theoretical machine used for computation and doing calculation very fast. The word “Automata” itself says it’s a automatic processing carried out for computation. Theory of automata is useful in study of how the machine doing the computation and solve the problem. Computation has been performed by processing the input by involving series of states. At each state a transition function determines the next configuration on the basis of a finite portion of the present configuration. As a result, once the computation reaches an accepting configuration, it accepts that input. The most general and powerful automata is the Turing machine. The motivation behind the automata theory is to develop methods through which behaviour of the system can be analyzed. The theory of automata includes study of the Finite-state machine, Pushdown automata, Linear-bounded automata and Turing
machine. The families of automata above can be interpreted in a hierarchical form, where the finite-state machine is the simplest automata and the Turing machine is the most complex. A Turing machine is a finite-state machine yet the inverse is not true. [3]

In 1930s Alan Turing studies Turing machines. During the year of 1940-1950s Finite automata machines have been studied and also the chomsky Hierarchy for formal languages have been proposed by Noam Chomsky. Then in 1969, Cook introduces “intractable” problems or NP-Hard problems.

Modern computer science, compilers, computational & complexity theory evolve in 1970s

Many researchers are still continuously working in the filed of theory of automata to understand the most complex systems and to develop many new system using automata. Some of the many new techniques which have been developed using automata are discussed in section 3.

3. Revolutionary Techniques in theory of automata

The basic principle of using automata theory to real world problems will remain intact. It can be applied to diversity of new field such as game designing, medical science in which used for medical diagnostics, inventory management, business and strategic planning.

In this section following four new techniques have been discussed:

3.1. Automata in XML
3.2. Automata in Game Designing
3.3. Automata in Biology
3.4. Automata in P vs NP Problems

3.1. Automata in XML

Since the arrival of XML as a data representation language, concepts from formal language theory like regular expressions, grammars and automata have been used for various purposes, e.g., as algorithm models for efficient evaluation of simple queries, as a proof tool, as a tool for static analysis and as an operational model with a clear semantics. Besides automata that read XML documents as strings, called document automata in this article, tree automata play an important role, as XML documents have a tree structure and for many applications it is possible to abstract away from text and data details and to model XML data by labelled trees. [4]

3.1.1. XML Documents as Trees

For most theoretical purposes, XML documents can be adequately modeled finite unranked, ordered, labelled trees with additional data values. In first place, an XML document is a sequence of symbols, i.e., a string. The structure of valid XML documents is described in [5] by a combination of, basically, an extended context-free grammar with additional constraints. Figure 1 (a) shows an example of a (toy) XML document. [4] As they are strings, XML documents can, in principle, be handled by string automata (called document automata below). Nevertheless, for reasons discussed below, most of the investigations covered in this survey exploit the intrinsic tree structure of XML documents and employ tree automata instead of string automata. Thus, we discuss in this section several approaches to represent XML documents as trees. [4]

Figure 1: Example document (a), as a tree (b), as a term (c), the underlying tree domain (d), its label function (e), its binary encoding via first-child-next-sibling (f), as in [6] [4]

The most direct way of representing an XML document as a tree is illustrated in Figure 1 (b).

- Elements and text strings of the document become nodes of the tree.
- Whenever an element or a text string is directly contained in another element, there is an edge from the node representing the containing element to the one representing the content. Thus, nodes representing character data are always leaves of the tree (but not vice versa).
- Additionally, the children of a node are ordered from left to right, corresponding to the order in the
document. Therefore, as opposed to usual graph presentations, the order in which the children of a node are depicted is important.

- Finally, in the tree representation, element nodes are labeled by the respective element name. [4]

### 3.2. Automata in games design

Automata tools have become widely used in the designing and development of computer games and computing game theory. There is a wide scope of research in this field. Deterministic finite state automata (DFSA) and non-deterministic finite state automata (NDFSA) are used primarily in various levels of designing the game. The designing of a game includes designing of various levels, selection, movement, action, game sets etc. Automata is divided to design all these features. Each feature is described as a state of automata. States are assigned labels or tags. DFSA and NDFSA tools are used to move from one level to other level and to get the state. Non deterministic finite state automata cannot be directly developed using programming language. There is need to convert NDFSA to DFSA. Because in programming languages the epsilon edges in non-deterministic automata cannot be translated to commands. [7]

How can automata theory help to solve problems for games?

![Transition Diagram for game](image)

**Fig. 2 Transition Diagram for game** [7]

**Origin of Game Problem:**

Circuit synthesis and Church’s problem (1957)

**Setting:** [8]
- Sequence of input signals arrives
- Circuit produces a sequence of output signals (depending on the inputs it has seen)
- Result is a non-terminating sequence of input and output signals
- A logical specification describes the desired properties of these sequences

![Input Output circuit diagram](image)

**Fig. 3 Input Output circuit diagram** [8]

**Problem:** Decide if there is a sequential transformation \( f: \Sigma_1 \to \Sigma_2 \) realizing \( \phi \), and construct one if possible.

Modelled as a game:

- One player plays input signals, the other player output signals
- The specification is the winning condition for the output player
- A transformation \( f \) realizing the specification is a winning strategy for the output player. [8]

**NDFA To DFA:** Deterministic Finite Automata is a real machine but non-deterministic finite automata is not a real machine and it is easier to maintain the non-deterministic property inspite of deterministic finite automata which is ideal to deal with programmed in any computing language. So to translate the non-deterministic finite state automata (NDFSA) and minimization may be applied using Kleene's theorem of unification to obtain the regular expression which may be translated then to deterministic finite state automata. [8]

### 3.3. Automata in biology

What Is MeantBy “Biological Computation”?

The term biological computation refers to the proposal that living organisms themselves perform computations, and, more specifically, that the abstract ideas of information and computation may be key to understanding biology in a more unified manner. It is important to point out that the study of biological computation is typically not the focus of the field of computational biology, which applies computing tools to the solution of specific biological problems. Likewise, biological computation is distinct from the field of bio-inspired computing, which borrows ideas from biological systems such as the brain, insect colonies, and the immune system in order to develop new algorithms for specific computer science applications. While there is some overlap among these different
moldings of biology and computer science, it is only the study of biological computation that asks, specifically, if, how, and why living systems can be viewed as fundamentally computational in nature. [9]

Automata Application in Molecular biology:

It is the study of biology on a molecular level including the structure, function, and makeup of biologically important molecules such as DNA, RNA, and proteins. Is the branch of biology that deals with the formation, structure, and function of macromolecules essential to life, such as nucleic acids and proteins, and especially with their role in cell replication and the transmission of genetic information.[10]

Transfer of information from protein to protein or from protein to nucleic acid is impossible. All cells, from the simplest bacteria to humans, express their genetic information in this way - a fundamental principle of this dogma. The finite automaton below represents the central dogma: [10]

![Central Dogma Diagram](image)

There are special cases of transfers of biological sequential information. Reverse transcription, RNA replication and direct translation from DNA to protein. The special transfers were indicated in the automata above (red arrows). [10]

Traditional Versus Biological Computing

In my view, to make the notion of biological computation more precise in any particular system, we need to answer the following questions (Mitchell, 2009):

- How is information represented in the system?
- How is information read and written by the system?
- How is it processed?

- How does this information acquire function (or "purpose", or "meaning")?

These questions are relatively straightforward to answer for traditional computing systems, which are based on the Turing-machine and von Neumann-style architectures. In such systems information is represented as collections of bits, which represent components of programs or data. Information is read and written by a central processing unit via "fetch" and "write" operations to and from memory. Information is processed via logical operations performed by the CPU. This information acquires "meaning" via the interpretations of human users.

The processing of information in traditional computers is centralized (i.e., performed by a CPU), typically serial, deterministic, exact, and terminating (i.e., there is an unambiguous final result of the computation). On the other hand, in biology, information processing is massively parallel, stochastic, inexact, and on-going, with no clean notion of a mapping between "inputs" and "outputs".

Whereas traditional computing systems typically require synchronization in many aspects of their processing, biological systems often operate with asynchronous components. Traditional computing systems require components to be reliable, with very low error probabilities, whereas biological systems operate with unreliable components that are subject to frequent failures. In traditional computer science, the notions of universal computation and programmability are fundamental, whereas the relevance of these concepts for biological computing is unclear. [11]

3.4. Automata in P Vs NP Problem

An approach is existed for P vs. NP problem through algebraic geometry.

**Problem:** Hamiltonian Path & Hamiltonian Cycle. If the graph contains the Hamiltonian cycle then it must contains Hamiltonian path, but converse cannot be true.

Hamiltonian path problem cannot solve in polynomial time (P), but we can verify in non-deterministically polynomial time (NP).

Let for some n, if Pn contains an integral point, then any Hamiltonian cycle for the Hamiltonian path problem must have size at least n^{\log n} on inputs of size n, which means P ≠ NP. Then to show that P ≠ NP it must satisfies that Pn contains an integral point for all n. [12]
It is necessary to show that $P_n$ contains an integral point for all $n$, first showing that the integer programming problem for $P_n$ is in $P$. For this approach, there are three steps:

1. Prove that the integer programming problem for $P_n$ is solve in polynomial time,
2. Find an simple, efficient algorithm for the integer programming problem for $P_n$, and
3. Prove that this simple algorithm always provides the answer is ‘Yes’.

Since the polygons $P_n$ are algebro-geometric in nature, solving (1) is thought to require algebro-geometric geometry, representation theory, and the theory of quantum groups. These conditions have classical analogues that are known to hold, based on the Riemann Hypothesis over finite fields (a theorem proved by Andre Weil in the 1960s).

**P vs NP Status:**

- This survey focused on the P versus NP problem, its importance, our attempts to prove $P \neq NP$ and the approaches we use to deal with the NP-complete problems that nature and society throws at us.
- Much of the work mentioned required a long series of mathematically difficult research papers that we could not hope to adequately cover in this short article. Also the field of computational complexity goes well beyond just the P versus NP problem that we haven't discussed here.
- The P versus NP problem has gone from an interesting problem related to logic to perhaps the most fundamental and important mathematical question of our time, whose importance only grows as computers become more powerful and widespread.
- Proving $P \neq NP$ might not be the start of the story either. None of us truly understand the P versus NP problem; we have only begun to peel the layers around this increasingly complex question. Perhaps we will see a resolution of the P versus NP problem in the near future but I almost hope not. [12][13]

**Conclusion**

Many revolutionory techniques such as desiging of games, automata used in XML, biology, neural network and decideability like P vs NP problems have been discussed in this paper. The motivation behind this paper is to give an overview of some of the new techniques of automata to open the new research direction in this filed.

**References:**

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Abstract: This paper uses ANSYS Product Software a Review to solve for the deflections and reaction forces for the Composite helical spring. This paper also is intended only as an educational tool to assist those who wish to learn how to use ANSYS and help for determining suitable modeling methods for any application. By considering suitable diagrams, concern images we will model and analyze a static, loaded spring system. The adoption of Finite Element Method, with Concern instructions also used to easily understand which promote the researchers o further Research in the field of analyzing their concepts

Keywords: Composite helical spring, Loading, Ansys Prod., Analysis etc.

INTRODUCTION:

Modified parameters in industries to forcing the companies to closely examine the way in which they develop products but condition is that they must be formed from main characteristic parameters. It has different type of applications in different areas. According to that the design considerations are to be made which discussed by every Researcher. The future discussion about basic phenomenon’s like stability of spring, surge in spring, spring relaxation, fatigue loading, strain energy and basic design procedure of the helical springs must be considered.

We consider the Pre-processing and Post-processing instructions. They include alternative command line entries that can be ignored if you choose to use menu picks to perform the required tasks. These commands are provided for our information. We may find that it is sometimes more convenient to enter certain commands directly at the command line.

We consider the Pre-processing and Post-processing instructions. They include alternative command line entries that can be ignored if you choose to use menu picks to perform the required tasks. These commands are provided for our information. We may find that it is sometimes more convenient to enter certain commands directly at the command line.

Steps Involved:

The following steps are considered to run the analysis on “ansys Product”

1. Alter Job Item/ Job Name
2. Define element type i.e. Helical spring-damper element
3. Define real constants. i.e. Spring Constant, or “Stiffness
4. Create K_I parallel K_II
5. Create K_II in Series with result of K_I, K_III
6. Create nodes. (‘four’ total)
7. Create first spring element between nodes ‘one’ and ‘two’.
8. Alter real constant set number to ‘two’ i.e. relates to the
9. helical spring constant used for elements
10. Create the remaining two helical spring elements.
11. Apply constraints and loads to this model
12. After Loading Find values
13. Plot deformed shape
14. Plot Unreformed shape if exist
15. List all forces
16. List reaction forces
17. List the deflections at each node
18. Exit the ANSYS program

Command of processing:

There is a complete listing of the alternative command line entries immediately used f. The entire analysis can be run by entering all of these commands at the command line, in the order shown. ANSYS also has the ability to read in a text file
containing these commands. Such a file would be called a “batch” file. The command list can be stored in a text file, and then read into ANSYS. One way to do this would be to store the file in your ANSYS working directory. The, in the ANSYS Graphical User Interface, select (top left of the GUI):

Then, browse to find the correct file, and double click on the file’s name. ANSYS would then execute the commands in the file in sequence.

Rigid-bar having unit in lb/In

Take value of $K_r=1000$,
Take value of $K_{II}=2000$,
Take value of $K_{III}=10000$,
Take value of $F_x=5000$

Command Input:

```
/filnam, spring
/prep7
/et,1,14
Combin14
r,1,1000
1, VALUE k2=2000 (spring constant)
r,2,500
2, VALUE k1=1000 (spring constant)
n,1,0,0,0
(x,y,z)=(0,0,0)
n,2,1,0,0
(x,y,z)=(1,0,0)
n,3,2,0,0
(x,y,z)=(2,0,0)
n,4,2,0,0
(x,y,z)=(2,0,0)
nlist
!List the nodes and node locations
/pnum, node, 1
!Specify jobname
e, 1,2
!Set plotting option so that
element with end nodes 1 and 2.
Real,2
!Define a Combin14
e,2,3
constant set to set number 2 (k3=1000).
e,2,4
!Define a Combin14
e,2,4
element with end nodes 2 and 3.
/solu
d,1,all,0
!Set the active real
to zero.
d,3,all,0
to zero.
d,4,all,0
to zero.
d,2,uy,0
to zero.
d,2,uz,0
to zero.
/so
f,2,fx,-5000
!Apply a force of
magnitude 5000 lb. to node 2 in negative x-dir.
solve
/post1
prnl
!Print the nodal
displacements.
prns
!Print the nodal displacements.
save
!Save all information to a
binary file named “spring.db”
/eof
!Exit ANSYS
```

Pre-processing:

- **Alter jobname:** File -> Change Jobname
  Enter “Helical spring”, and click on “OK”.
  Other Command Line Entry = /filnam, spring
  Also, to enter the preprocessor, at the command line, enter:
  `/prep7`

- **Define element types:** Preprocessor -> Element Type -> Add/Edit/Delete
  Click on “Add”, highlight “Combination”, then “Spring-damper 14”: 

```
Experimentally for straight beams and helical springs. In their study, the effects of the number of active loops, the helix

Enter “Helical spring”, and click on “OK”.
Other Command Line Entry = /filnam, spring

Also, to enter the preprocessor, at the command line, enter:
/prep7

Define element types: Preprocessor -> Element Type -> Add/Edit/Delete

Click on “OK”, then “Close”. Note that in ANSYS this element is sometimes referred to as “Combin14”, because it is element type 14 in the ANSYS element library, and can be used for both stiffness and damping in this model. For a static analysis, damping has no effect. (Other Command Line Entry = et,1,14)

Define the real constants for the Combin14, which for this is only the spring constant: We need two real constants sets, because there are two different spring constants in the system.

Preprocessor -> Real Constants -> Add/Edit/Delete
Click “Add”, then “OK” for “Type 1 COMBIN14”

After filling in the spring constant value, click on “APPLY”. Then, change the real constant set number to 2, and enter a spring constant value of 2000. Then, click on “OK”, then click on “CLOSE”.
Other Command Line Entry = r,1,2000
Other Command Line Entry = r,2,500

Create nodes: Preprocessor -> Modeling -> Create -> Nodes -> In Active CS
Enter 1 for node number (ANSYS would automatically number nodes if you left this blank).
Enter the location as (X,Y,Z)=(0,0,0). Note that we are entering the locations in a Cartesian coordinate system. Leave the entries for rotation angles blank.
For this problem, all nodes will be on the X-axis, with Y=0 and Z=0.
Click on “Apply”. Define node 2 at (X,Y,Z)=(1,0,0), and click on apply, then define node 3 at (X,Y,Z)=2,0,0, and click on apply, then define node 4 at (X,Y,Z)=2,0,0, and click on OK. These X-locations are somewhat arbitrary, as the results do not depend on the distance between nodes for the Combin14 element type.

Other Command Line Entry = n, 1,0,0,0
(or, simply: n,1; missing input is interpreted by ANSYS as “zero” in this case).

Other Command Line Entry = n, 2,1,0,0
Other Command Line Entry = n, 3,2,0,0
Other Command Line Entry = n, 4,2,0,0

Click “OK” and the list will appear.

Turn on node numbering:

Utility Menu -> PlotCrls -> Numbering.

Check “on” for “node numbering”, then click “OK”. The node numbers may already be showing, but this will force the display of node numbers on subsequent plots.

Other Command Line Entry = /pnum, node, 1

Create a spring element between nodes 1 and 2:

Preprocessor -> Modeling -> Create -> Elements ->Auto Numbered->Thru Nodes

• Change the real constant set number to 2.

Preprocessor -> Modeling -> Create -> Elements -> Elem Attributes
Modify the “Real constant set number” to 2: Then, click on OK. (Other Command Line Entry =real, 2)

• Create the remaining two spring elements:
Create two more elements, one from node 2 to node 3, and one from node 2 to node 4. This is a little tricky because nodes 3 and 4 are coincident. To create the element between nodes 2 and 3, choose:

Preprocessor -> Modeling -> Create -> Elements ->Auto Numbered->Thru Nodes

Other Command Line Entry = e,2,3
Other Command Line Entry = e,2,4

If entering commands at the command line, to enter the solution processor, type: /solu

• Apply constraints and forces on the model:

Solution -> Define Loads-> Apply -> Structural-> Displacement -> On Nodes
A picking menu appears. Pick node 1, then pick node 3, and then pick node 4. When all three are selected, the picking menu will appear as below, where “Count=3” and “Maximum=4”:

Other Command Line Entry = d, 1, all, 0
Other Command Line Entry = d, 3, all, 0

Find Values: Solution -> Solve -> Current LS
Click “OK” in the “Solve Current Load Step” Box.
Other Command Line Entry = solve

Post-processing: If entering commands at the command line, to enter the postprocessor, type: /post1:
Plot the deformed shape
General Postproc -> Plot Results -> Contour Plot ->Nodal Solution
On the box that opens, choose:
DOF Solution -> X-component of displacement
Click “OK

- 323 -
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"Lingaya’s Group of Institutions" was established in the memory of the freedom fighter “Late Shri Lingaya Gadde”. Shri. Lingaya, an agriculturist by profession took part in the 1923 civil disobedience and quit India movement and went to jail several times, besides this he wrote several books on socio-communist issues. He was an editor of the "Navayuga Communists Party Organizer and also the founder of the "Adarsa Grandha Mandal", Elamaru, Andhra Pradesh. The society of “Lingaya's Jankalyan Shikshan Sanstha” has its foundation with thoughts of social development and its walls are made up of devotion, sacrifice, sanctity of feeling and candid thoughts, thus making it strong to achieve those dreams of the freedom fighters.

Prof. G. V. K. Sinha, an educationist with remarkable insight into technical education and great expertise in managing the human resources was a self-made icon in the field of technical and management education. He started his career as a lecturer, climbed up the ladder as head of the department, Asst. Director and then the Principal in various reputed technical institutions in New Delhi; and later full time consultant of AICTE, New Delhi. Due to his vision, perseverance, grace and humanitarian values Prof. Sinha finally reached to the peak of his career as the Chairman of Lingaya’s Institute of Management and Technology (LIMAT), Faridabad set up by him in memory of his father, the great freedom fighter Gadde Lingaya. LIMAT is the living legend of Prof. Sinha’s noble goal and higher vision in the field of education. His ultimate dream was fulfilled when the institution set up by him was declared as Deemed-to-be - University. On 6 June 2011, Prof. Sinha left us for the heavenly abode. May the departed soul rest in peace! Let us pay tribute to Prof. Sinha with a pledge to uphold and keep up his spirit and vision on education.

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