Dr. V.S.K. Reddy, Conference Chair

Dr. V.S.K. Reddy, Principal, Malla Reddy College of Engineering & Technology, has an experience of 22 years in Teaching and Research put together. He is an alumni of IIT Kharagpur and obtained Ph.D in the area of Multi-media Signal Processing and Communication Protocols. He is versatile in multidisciplinary specializations in Electronics & Communications and Computer Science Engineering. His laurels include more than 100 Publications in the National and International reputed Conferences and Journals. He is a fellow of IETE, Life Member of ISTE and Member of IEEE. He was awarded as "Best Teacher" for three consecutive Academic years with citation and cash award. He is the recipient of "India Jewel Award" for outstanding contribution in research in the field of Engineering and Technology. He is the Member of Board of studies for M.Tech program Sreenidhi Institute of Science & Technology, Hyderabad collaborating with M/s. Synopsis-SEER Academy, USA. He is also a member of Board of Studies for ECE & ETM, JNT University, Hyderabad, India.

Prof. P. Sanjeeva Reddy, Convener

He completed his B.E (Tele-Comm.) from Osmania University in 1966 and M.Tech, from IIT Madras in 1979. He has 36 years of experience in **DRDO** as a scientist in various laboratories out of which 20 years as a radar scientist at LRDE, Bangalore. He developed three radar systems out of which two radars viz. Indigenous Doppler Radar (INDRA) & Distance Measuring Equipment (DME) were successfully inducted into the Services. He is the recipient of Republic Day award from Director, LRDE for outstanding contribution. For the next 16 years he worked as scientist in DRDL and RCI, Hyderabad and was actively associated with the development of a short range surface to air missile system under the guidance of Dr. APJ Abdul Kalam, former President of India and later served as Director, Advanced Missile Technologies at RCI. He was also deputed as visiting scientist to M/s.Selenia Co., Rome, Italy for two months. He retired from the DRDO service in Oct.2005 and joined as Professor in ECE at MRCET. Presently he is Director, School of Electronics and Communication Engineering and Dean for International Studies. He is a member of IEEE and Life Fellow of IETE.

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on

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December 16-17, 2016

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INTERNATIONAL CONFERENCE ON COMMUNICATIONS, SIGNAL PROCESSING, COMPUTING AND INFORMATION TECHNOLOGIES

(ICCSPCIT-2016)

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COMMUNICATIONS, SIGNAL PROCESSING, COMPUTING AND INFORMATION TECHNOLOGIES

Edited by

Dr. V S K Reddy

Malla Reddy College of Engineering & Technology

Maisammaguda, Secunderabad – 500100

Prof. P. Sanjeeva Reddy

Malla Reddy College of Engineering & Technology

Maisammaguda, Secunderabad – 500100

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In Pursuit Of Knowledge

Message from the Chief Guest

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Dr. C.G. BALAJI
Distinguished Scientist
Director

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I am happy to know that the Malla Reddy College of Engineering & Technology, Maisammaguda, Dhulapally, Secunderabad is organizing the fifth International Conference on Communications, Signal Processing, Computing and Information Technologies ICCSPCIT-2016 on 16-17 December, 2016. The prime responsibility of shaping upright and worthy citizens of tomorrow is the pivotal role of today's premier educational institutions such as MRCET. Education today is global in perspective and practice. Both faculty and research scholars have to keep themselves abreast of international technological developments in their area of specialization as well as in the related fields. I am sure that the international conference will provide the necessary forum for all the participants and delegates to share and enrich their knowledge.

I wish the International Conference all the success in meeting its objectives.

Yours sincerely

Message from Chief Patron

I believe Education is transmission of civilization. I visualize that the students of MRCET to be sound in technical knowledge as well as enterprising. MRCET has become a trend setter for MRGI. The college has continuously endeavored to groom dynamic leaders mingling with both academics and extracurricular activities. It's a matter of great pride that not only in academic but also in sports activities MRCET has become a "Force to reckon with".

I am happy to know that Malla Reddy College of Engineering and Technology is Organizing the fifth International Conference on Communications, Signal Processing, Computing and Information Technologies ICCSPCIT-2016 on 16-17 December, 2016. It is indeed a very prestigious event for MRCET and I wish the conference a great success.

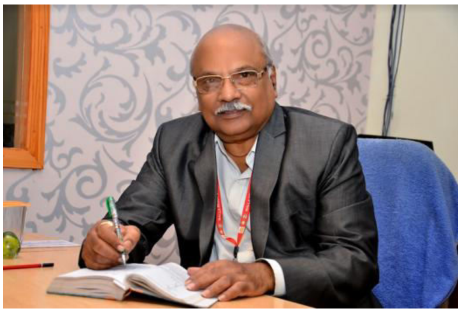
Sri. Ch. Malla Reddy,
Founder Chairman

Message from Conference Chair

It is a great pleasure for me to announce that Malla Reddy College of Engineering and Technology is Organizing the fifth International Conference on Communications, Signal Processing, Computing and Information Technologies ICCSPCIT-2016. To meet the innovative standards in engineering research there is a dire necessity to boost up the quality of engineering education at a rapid pace to catch up with the rapidly changing technology. The previous four conferences met with great success in terms of both the quality and quantum of papers presented and I strongly believe that this conference will also provide an excellent opportunity to all the professionals, academicians and researchers working in the areas of Communications, Signal Processing, Computer Science, and Information technologies to share their knowledge, experiences and expertise and thereby contribute to the growth of technology. I wish the conference all the success.

Dr.VSK Reddy,
Principal

Message from Convener

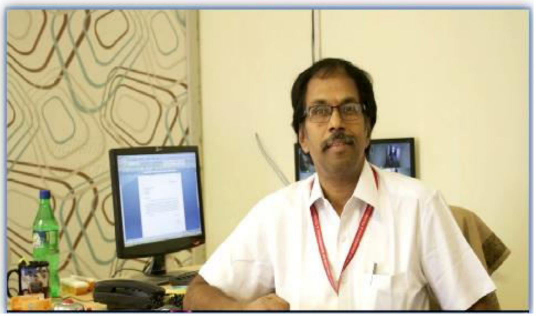


It is a great honour for me to be the convener for the fifth International conference on Communications, Signal Processing, Computing and Information Technologies ICCSPCIT-2016 focusing on the areas of Communications, Signal Processing, Computing and Information technologies. The response of authors for the fourth International Conference ICCSPCIT-2015 has been amazing and I am earnestly hoping for similar response for this conference also. It is evident that there is a pressing need for both faculty and professionals from the industry to keep abreast of the latest technological developments and it is the prime objective of the international conference to provide the necessary platform for such information interchange among all the participants for sharing of knowledge and expertise.

I wish the international conference ICCSPCIT-2016 a great success in fulfilling its objectives.

Prof.P.Sanjeeva Reddy,
Dean, International Studies

Message from Organizing Secretary



Overwhelming response and support from the technology community for ICCSPCIT 2015 have inspired us to organize ICCSPCIT-2016, the 5th International Conference on Communications, Signal Processing, Computing and Information Technologies (ICCSPCIT - 2016) at Malla Reddy College of Engineering & Technology (MRCET), Hyderabad, Telangana, India on December 16 & 17, 2016. This conference aims to integrate the research, scientific and industry communities. I invite all of you (academician/researchers/industry professionals /sponsorers and exhibitors) working in different areas to participate and attend the event and make it a grand success. ICCSPCIT has grown in this part of the world as perhaps the most comprehensive technical forum for sharing the latest outcomes of research and developments to industrial applications in the areas of Communications, Signal Processing, Computing and Information Technologies. Considering the timing and place of the event (growing economy), ICCSPCIT-2016 provides unique opportunities for academic, research and business collaborations.

On behalf of the organizing committee, I welcome you all to ICCSPCIT-2016 at MRCET, Hyderabad, Telangana, India on December 16-17, 2016 for this exciting event.

Dr S.Srinivasa Rao
Organizing Secretary
Professor& Head, ECE

Message from Organizing Secretary

From the Department of Computer Science and Engineering, I am indeed privileged and delighted in hosting the Fifth International Conference on Communications, Signal Processing, Computing and Information Technologies (ICCSPCIT 2016), on 16th and 17th December 2016.

Organizing such an event at this point of time reinforces our objective of developing an environment for the exchange of ideas towards technological developments. I wish the conference would be able to deliberate on current issues of national and international relevance, particularly in the field of cloud computing, data mining, networks, image processing, big data analytics etc. I am sure that this occasion will provide an affable environment for the researchers and academicians to freely exchange the views and ideas with others.

I express sincere thanks to the Respected Chairman Sri. Ch. Malla Reddy garu and beloved Principal Dr. VSK Reddy garu for providing the platform to organize ICCSPCIT-2016. Such a herculean task would not have been possible without the voyage traveled together by the organizing committee members, National and International academicians who have traveled far distance for their participation and researchers.

Dr. D .Sujatha
Organizing Secretary
Professor& Head, CSE

Message from Organizing Secretary

It is a great privilege to inform that Malla Reddy College of Engineering Technology conducting 5th international conference on Communications, Signal Processing, Computing and Information Technologies (ICCSPCIT-2016).

I strongly believe, such conferences bring researchers, engineers and faculty members to single platform to exchange their knowledge and to improve their communicative, competitive and innovative skills.

I wish for grand success of the event.

Prof.T.Prakasam
Organizing Secretary
Professor& Head, IT

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Foreword

Malla Reddy College of Engineering and Technology (MRCET), is a constituent college of Malla Reddy Group of Institutions (MRGI). The college was established in the year 2004, approved by AICTE, New Delhi and is affiliated to JNT University, Hyderabad and is also an UGC Govt. of India Autonomous Institution. The College is offering B.Tech and M.Tech Programmes in the areas of Electronics & Communication Engineering, Computer Science and Engineering, Mechanical Engineering, Aeronautical Engineering, Information Technology, Mining Engineering and MBA. The college is equipped with state of the art laboratories for all the departments, full–fledged training & placement cell and an Industry Institution Partnership Cell. The college is NBA accredited and also NAAC accredited with ‘A’ grade. The college is an ISO 9001:2015 certified Institution.

In the global competition, quality has become the essence of the success in every walk of life. To meet the innovative standards in engineering research there is a dire necessity to boost up the quality of engineering education at rapid pace to catch up with rapidly changing technology. This is a dynamic situation and we have to continuously update the expertise in various fields of specialization. Research activities of this kind will help to ignite the young brains of the institution which will lead to a meaningful and innovative research. Malla Reddy College of Engineering and Technology is organizing the fifth International Conference on Communications, Signal Processing, Computing and Information Technologies ICCSPCIT-2016 on 16-17 December, 2016. I am happy to mention that the previous four conferences met with great success in terms of both the quality and quantum of papers presented. The latest trends and emerging technologies in Engineering play a vital role in both industry and research. Creation and dissemination of knowledge is sacred mission of any technical institution and we are sure that this two-day International Conference will provide an opportunity for the Academicians, Researchers and Engineers working in this area to exchange their ideas, knowledge and have fruitful discussions which will provide an impetus for the rapid familiarization and advancements in the fields of specialization.

We express our sincere thanks and gratitude to our visionary Chairman Sri Ch. Malla Reddy for constant support and encouragement for organizing this International Conference. We thank all technical session chairs and both National and International experts in our Advisory committee for their valuable suggestions and guidance in organizing our conference.

Conference Chair
Dr. VSK Reddy

Conference Theme

The International Conference on Communications, Signal Processing, Computing and Information Technologies (ICCSPCIT – 2016) is planned to create awareness and provide a common platform for the professionals, academicians and researchers working in the area of Computing, Communications, Signal Processing, Information Technologies and System Design to analyze state of the art developments, innovations and future trends and thereby contribute to the much needed dissemination of latest developments and advances in the Field of Engineering & Technology. It is a well known fact that research plays a vital role in the sphere of teaching and academics. Signal Processing and Communication Engineering, encompasses topics such as VLSI, Embedded Systems, Wireless and Mobile Communications, Image and Video Processing, Microwave and Radar Engineering being the State of art subjects which have exhibited rapid growth and have numerous and wide spread applications.

Computing and Information Technologies cover a wide range of topics ranging from Data Mining, Image Processing, Internet of Things, Cloud and Grid Computing and among many others. The various intelligent tools like swarm Intelligence, Artificial Intelligence, and Evolutionary Algorithms have been applied in different papers for solving various challenging IT related problems. This conference provides a platform for interacting and exchanging ideas, experience and expertise in the current focus of global research, recent developments, challenges and emerging trends in the above mentioned fields. Finally, ICCSPCIT 2016 is going to be an intellectual arena which encompasses ability to exchange and exhibit mutual respect and understanding of ideas, philosophical principles, cognitive styles, and mind-sets as well as acts of integrity and purposeful personal reflection.

The Submitted papers are expected to cover state-of-the-art technologies, product implementation, ongoing research as well as application issues.

CONTENTS

THE FUTURE OF CARTOGRAPHY SATELLITES DATA ACQUISITION AND ITS DATA INGEST SYSTEMS

DR.A.N.SATYANARAYANA[A] B.CHANDRASEKHARA RAO[B] DR.YVN KRISHNA MURTHY[C] K.V.RATNA KUMAR[D]

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National Remote Sensing Centre, Indian Space Research Centre, Dept.of Space, Govt. Of India
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ABSTRACT: Remote sensing data provides much essential and critical information for monitoring many applications such as image fusion, change detection, and land cover classification. Remote sensing is an important technique to obtain information relating to the Earth's resources and environment. What popularized satellite data are the easily accessed online mapping applications like BHUVAN. From being simply able to find "where is my house" these applications have helped the GIS community in project planning, monitoring disasters and natural calamities, and guiding civil defence people. ISRO's BHUVAN (www.bhuvan.nrsc.gov.in) is a well known national geo-portal, which is being widely, used by the Government, public, NGOs and Academia. Bhuvan is developed with a clear focus of addressing Indian requirements of satellite Images and theme-oriented services to enable planning, monitoring and evaluation of stakeholder's activities in governance and development. All these data is taken from the Remote sensing satellites of Cartography application series Satellites, which are being tracked and data ingested disseminated from IMGEOS, NRSC Ground Station. The acquisition challenges and data ingest process will be discussed in this paper.

KEY WORDS: IMGEOS: Integrated Multi mission Ground Station For Earth Observation Satellites, BHUVAN: Is a well known national geo-portal, Antenna Systems, Tracking Systems, SERVO, RF Systems.

1. INTRODUCTION

1.1 Data Acquisition from Cartography Satellites

There are hundreds of applications for satellite imagery and remotely sensed data. From the Indian remote Sensing satellite Series, nations used to use information derived from the satellite imagery for spying on each other under the guise of scientific experiments, industry has grown in leap and bounds and today every sphere of life, government decision making, civil defence operations, police, you name the sphere of life, every one of which is influenced by satellite imagery in particular and Geographic Information Systems (GIS) in general.

Earth Observation by Remote Sensing Satellites [1] by NRSC began by commissioning an integrated facility with Data acquisition and Product generation at Shadnagar for Landsat MSS data in 1979. This was followed by development of LANDSAT-TM data processing and establishment of SPOT data processing & product generation facility.

Development of IRS Spacecraft Technology Remote sensing is a multi-disciplinary domain and substantial progress has been made in several technologies during the past few decades. As the demand for "four Resolutions" – Spatial, Spectral, Radiometric and Temporal increase, the complexity of imaging sensors and in turns the satellite systems. While reliability calls for heritage systems, lessons learnt and the new requirements warrant new technology developments. A brief overview on these technology developments is given in the following paragraphs.

Data Reception and Archival is located at Shadnagar Ground Station complex and has four data reception terminals, to receive Image data in X-

Band and telemetry data in S band. These three terminals are configured to track and receive data from several National and international satellites (35-40) passes in a day). Configuration supports mission clashes and redundancy for important data reception. The Ground station has the capability to receive and demodulate the signals of up to 105 Mbps data rate. The feed and demodulators are upgraded for dual polarization and to cater for higher data rates. It has a Bore sight test facility to evaluate the total data reception chain for all three terminals.

Functionally data reception and auxiliary data processing consists of Antenna & Tracking pedestal, RF systems, Servo systems, Base band & Data acquisition systems and Ancillary data processing systems. Most of the sub-systems of antenna receive chain and interconnecting panels are not amenable for remote monitor and control. Hence they are to be manually set up for every data reception. There are Direct Archival & Quick Look Browse DAQLB systems for data archival and ancillary data processing from different satellites. Five of these support multi-mission operations for the current satellites. Operations of Data receive systems and DAQLB systems are manual.

Cartography Satellite, IRS-P5: Cartosat-1 is the first dedicated stereoscopic mission of ISRO, offering 2.5 m resolution in panchromatic band. Weighing 1560 kg at lift off, Cartosat-1 was launched into a 618 km Sun Synchronous Polar Orbit (SSPO) by PSLV-C6 on 5th May 2005. Cartosat-1 mission objectives are directed at geo-engineering (mapping) application, for high resolution panchromatic imagery with high pointing accuracies. The spacecraft features two high-resolution panchromatic cameras for in-flight stereo imaging.

2. BRIEF DESCRIPTION OF THE GROUND STATION FOR DATA RECEPTION FROM IRS-P5.

The ground station system configuration is explained with reference to the block diagram in Figure.1.2. The system consists of a diametric parabolic reflector antenna with cassegrain feed, mounted over an Elevation over Azimuth driven pedestal. The feed and front-end system realizes single channel monopulse signal tracking and data reception in X -Band frequencies. The sum and difference channel signals from the front-end system are fed to a five channel synthesized down converter are driven to the control room, wherein, after the amplitude equalization, the sum channel is fed to the data demodulation while the difference channel signal is fed to the tracking receiver. The data and clock signals from the demodulator and Bit synchronisers are fed to the archival systems during the pass. The tracking video output, corresponding to the antenna offset information in Azimuth and Elevation axes, from the tracking receiver is fed to the antenna control unit. The antenna control unit has several operational modes to control the antenna movement. The unit drives the antenna in auto track mode during the satellite pass with programme tracking mode operating as backup. The servo system [11] is a dual drive system with torque bias arrangements to avoid antenna backlash during tracking.

Initially the systems for receiving data from:

CartoSat-1 - formerly IRS-P5 (Indian Remote Sensing Satellite-P5)

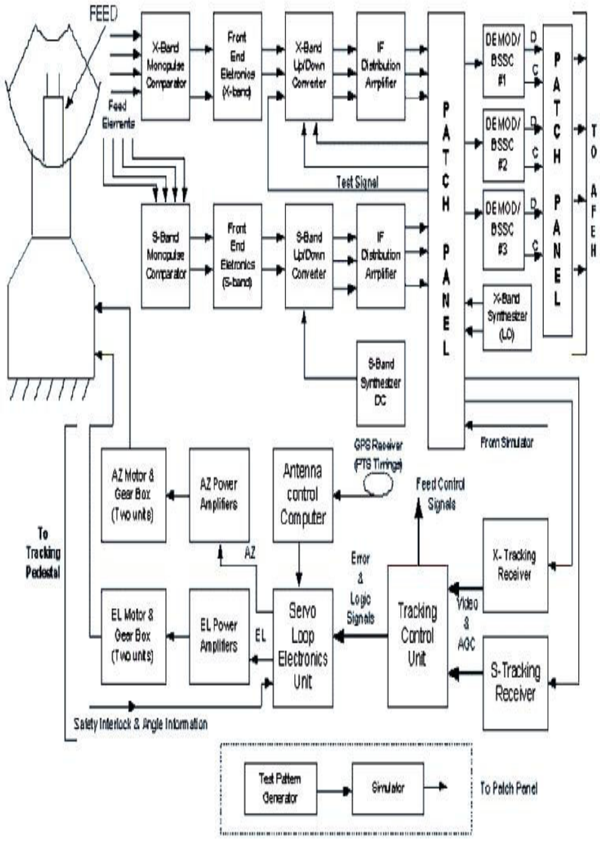

Fig.1.1 Ground Station Block Diagram

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IRS-P5 is a spacecraft of ISRO (Indian Space Research Organization), Bangalore, India. The objectives of the IRS-P5 mission are directed at geo-engineering (mapping) applications, calling for high-resolution panchromatic imagery with high pointing accuracies. The spacecraft features two high-resolution panchromatic cameras that may be used for in-flight stereo imaging. Prior to launch, ISRO renamed the IRS-P5 spacecraft to **CartoSat-1,** to describe more aptly the application spectrum of its observation data. In this mission, the high resolution of the data (2.5 m GSD) is being traded at the expense of multispectral capability and smaller area coverage, with a swath width of 30 km. The data products are intended to be used in DTM (Digital Terrain Model)/DEM (Digital Elevation Model) generation in such applications as cadastral mapping and updating, land use as well as other GIS applications.

2.1 Present Ground Station Scenario (IMGEOS)

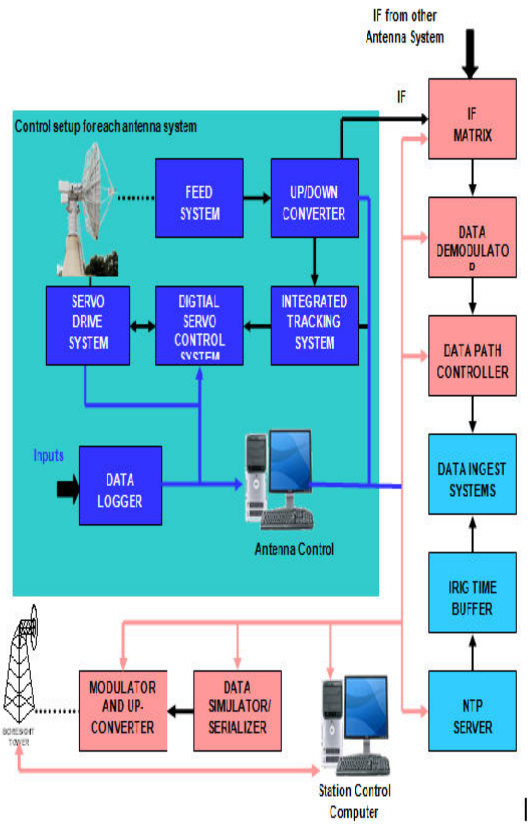

Fig.1.2 Present Ground Station Block Diagram

As the numbers of remote sensing missions have increased, and the scopes of supporting emergency requirements and disaster monitoring have increased, present mission specific model has several short comings. Hence there is a need to process re-engineer the entire chain, adopting an integrated multi-mission approach in order to minimize the satellite launch-to-product delivery and for improving the turn-around-time from data acquisition to product delivery to the user in near real time. To meet these objectives, an Integrated Multi-mission Ground segment for Earth Observation Satellites (IMGEOS) is proposed to be implemented at Shadnagar complex of NRSC.

The ground station configuration is shown in Figure 1-2. Four 7.5 meter antenna systems are used to track and receive data from different remote sensing satellites. There will be one antenna control computer (ACC) for each antenna systems and common station control computer (SCC) to perform the ground station operations.

Payload pass schedules & state vectors will be received through Sky-link on to SCC. After resolving the clashes, the SCC will assign the antenna systems for all scheduled passes after subjecting to acquisition strategies and archival policies. The SCC will configure base band systems automatically as per the schedules for all the four antenna systems and communicates its readiness to the ACC of the respective antenna system. The ACC in turn will configure the RF and IF sub systems before satellite pass tracking.

In real-time, the SCC and ACC monitor the required parameters from the base band and RF & IF systems respectively. The ACC will monitor all the parameters pertaining to RF & IF chain. In addition, it does program tracking which acts as a backup to the auto track mode. Subsequently it sends acquisition status to the SCC after the pass. During real-time, the Data Receivers log the data on to SAN based RAID system after doing demodulation and frame synchronization. The pre-processing system will provide the sub sampled quick look display in real-time. The station operations are planned to implement in fully automated environment aiming towards unmanned operations. The main objectives of the station automation are

- ✓ Visibility clash and elevation analysis
- ✓ Providing centralized control & configuration of the station
- ✓ Monitoring and control of the sub systems
- ✓ Building up of operational database

✓ Modularity for easy upgradeability to future missions

In multi-mission scenario, around 20 passes will be acquired covering both Indian and foreign satellites from four different Antenna Systems. These antenna systems will be operated simultaneously to acquire the data from different/ same satellites based on the clash scenario. In the operational scenario, it is required to reconfigure the chain and get ready for the next pass within 2 minutes. Around 25 parameters shall be monitored/ configured on various subsystems for each pass. Some of these important parameters are

Configuration parameters

- ✓ Local oscillator frequency and its output level of Down converter
- ✓ Gradient selection from Tracking servo control system
- ✓ Digital Phase shifter selections
- ✓ Routing of IF to different Data demodulators
- ✓ Demodulator, clock lock, polarity, data rate, output level, output mode

2.2 System Configuration

The configuration has been worked out keeping in view of the simultaneous operations of all four antenna systems to track different satellites and is shown in figure 1-3. Automation point of view, the total sub systems in the Data Reception System are divided into two categories- RF & IF systems (all the sub systems up to down converter in the data receive chain) and base band systems (from IF matrix onwards).

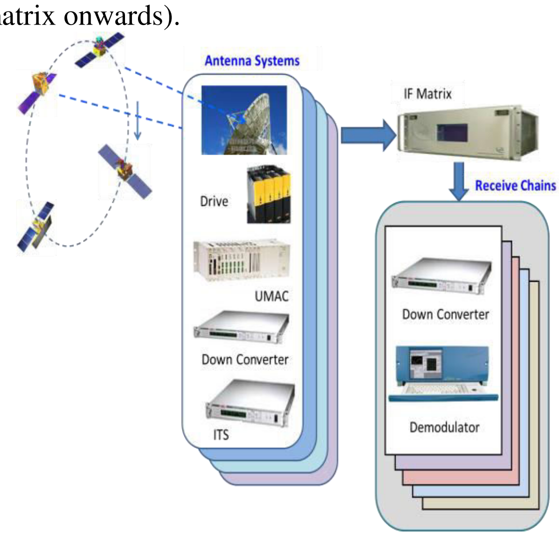

Figure 1.3 Station Automation Configurations

The RF & IF systems are dedicated to one antenna systems and there will not be any switching between the antenna systems in the operations scenario. The automation of these systems is carried out by the antenna control computer of the respective antenna system along with the program tracking of the satellite. Similarly, station control computer carries out the automation of base band systems. These systems include IF matrix, Data Demodulators, Data Simulators etc.

3.0 CartoSat -3 Series Satellites in Ka Frequency band

Indian Space Research organization (ISRO) is planning to launch next generation remote sensing satellites for cartographic applications, the first in the series being Cartosat-3 followed by Cartosat-3A & Cartosat-3B. These satellites transmit data to ground in Ka-band (25.5–27.0 GHz) and X band (7.8-8.5 GHz) with signals in Right Hand Circular Polarization (RHCP) and Left Hand Circular Polarization (LHCP) simultaneously. NRSC has the responsibility for acquisition of data from Cartosat-3/3A/3B Satellites, data processing, data product generation, dissemination as well as application development and capacity building.

The proposed antenna system is a 7.5m diameter shaped cassegrain type reflector with Monopulse feed assembly mounted on Elevation over Azimuth mount. Elevation over Azimuth Pedestal is equipped with Train axis. The three axes of rotation are arranged as elevation over azimuth over train with a 7o wedge (Tilt pedestal) between train and azimuth. Provision is made to orient the tilt in the designated direction anywhere in 360o as per the satellite pass requirements to avoid keyhole during overhead passes. The antenna is capable of ±360° in azimuth, -2° to 182° in Elevation and ±180° in Train axis coverage. This system is being under R&D.

3.3 Data Ingest

Station workflow manager will initiate data ingest operation for archival of satellite data based on Payload pass schedules. For each antenna systems one data ingest system is configured. One additional ingest system is also configured as standby. Ingested satellite data will be qualified for quality in terms of data losses. Data ingest systems in the context of ancillary data processing and storage systems are shown in figure 1-4.

Ancillary Data Processing

In near real time, the ancillary data processing systems (shown in figure 1-4) will generate ADIF,

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Browse and Histogram files for each pass. The generated ADIF will be populated on to the respective database for subsequent access and pre processing of data. The browse images are screened for quality & cloud automatically with a provision for manual certification before putting on to the Internet.

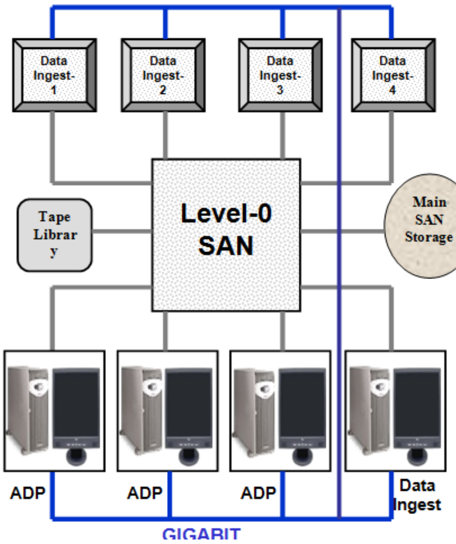

Figure 1.4 Ancillary Data Processing Systems Configuration

4.0 Storage Architecture

The Storage sub system is a Storage Area Network having an estimated storage of 25.0TB for acquisition & Level-0 processing, 15 TB for DP working area and around 720 TB of existing data of all satellites is archived on DLT/SDLT media (as on 31-Dec-2008)..

Data is stored online in 3 tier SAN storage. Provision is made for 100 TB (usable) accounting for three months of all satellites data in high performance storage. 400 TB (usable) accounting for 18 months data in medium performance SATA storage. All the data acquired and available archived data will be stored in Tape library in duplicate for backup and one more copy for vaulting. This 3 tier configuration is optimized for product generation and for data availability.

The following points are considered for selection of tiered storage

- ✓ For online processing or archival
- ✓ High performance vs. cost effectiveness
- ✓ Write once/ read many or continuous read/ write
- ✓ Frequently accessed or infrequently accessed
- ✓ File data or RDBMS data
- ✓ Heterogeneous platform access
- ✓ Multiple users or single user/ limited users
- ✓ Scenarios

- Data acquisition processes and data processing chains require high performance storage suitable for online processing etc.
- Archival data is mainly write-once read-many
- The past one year archival data is being accessed more frequently than earlier years' data.
- The storage is scalable to meet ever increasing data volumes with time. Storage architecture is shown in Fig. 1.5

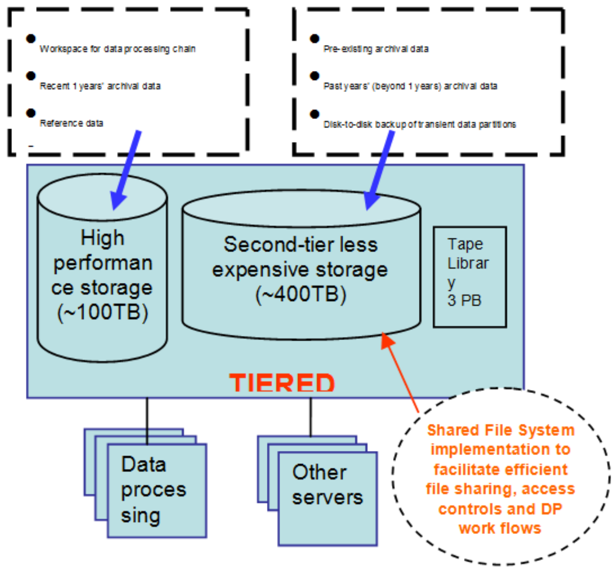

Figure 1-5 Storage Implementation-Main Storage

5.0 Data Archival for one day

Carto. Sat	Data Rate	Streams	Total Volume Data Acquired
Cartosat-01	105Mbps	Two	33.075
Cartosat-02	105Mbps	Two	14.175
Cartosat-2A	105Mbps	Two	3.938
Cartosat-2B	105Mbps	Two	33.075
Cartosat-2C	105Mbps	Two	33.075

CARTOSAT-3 TRANSMIT PARAMETERS IN Ka BAND

Orbit Height : 450 Km

Modulation Scheme : 8 PSK

Dual Polarization : RCP & LCP

Transmission Data Rate : 3.8 Gbps

6. CONCLUSION

The Advances in satellite communications technology in recent years have led to a significant increase in throughput delivered from a raft of new ‘High Throughput Satellite’ (HTS) systems. Carto 2 Series satellites have been launched in recent years and several more will go into orbit in the coming years. These satellites support diverse user requirements and use cases – from ‘connecting the unconnected’ to providing secure and resilient communications to industries. The satellite communications industry supports a wide range of customers whose varying use cases and deployment locations place exacting requirements that must be fulfilled. Connectivity must be delivered to consumers located beyond the reach of traditional terrestrial networks and yet also to business and corporate clients requiring bespoke systems to support critical communications needs

7. ACKNOWLEDGEMENTS

It is indeed a great pleasure my sincere gratitude to my centre Director Dr.Y.V.N.Krishna Murthy, NRSC and Dy. Director Sri.K.V.Ratna Kumar, for encouragement for producing this paper on future Satellite applications for Cartography applications.

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EFFECT & ANALYSIS OF QOS PARAMETER IN WIRELESS AD HOC NETWORK (IEEE 802.11B) FOR DIFFERENT PROTOCOLS

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***Abstract*—A wireless Ad-hoc network consists of wireless nodes communicating without the need for a centralized administration, in which all nodes potentially contribute to the routing process. A user can move anytime in an ad hoc scenario and, as a result, such a network needs to have routing protocols which can adopt dynamically changing topology. To accomplish this, a number of ad hoc routing protocols have been proposed and implemented, which include Dynamic Source Routing (DSR), ad hoc on-demand distance vector (AODV) routing, and temporally ordered routing algorithm (TORA).**

In this paper, we analyze the performance differentials to compare the above-mentioned commonly used ad hoc network routing protocols. We report the simulation results of four different scenarios for wireless ad hoc networks having thirty nodes. The performances of proposed networks are evaluated in terms of number of hops per route, retransmission attempts, traffic sent, traffic received and throughput with the help of OPNET simulator. Channel speed 11Mbps and simulation time 20 minutes were taken. For this above simulation environment, TORA shows better performance over the two on-demand protocols, that is, DSR and AODV.

***Keyword*— AODV, DSR, TORA, OPNET, MANET, DSSS**

I. INTRODUCTION

A wireless Ad-hoc network consists of wireless nodes communicating without the need for a centralized administration. A collection of autonomous nodes or terminals that communicate with each other by forming a multihop radio network and maintaining connectivity in a decentralized manner is called an ad hoc network. There is no static infrastructure for the network, such as a server or a base station. The idea of such networking is to support robust and efficient operation in mobile wireless networks by incorporating routing functionality into mobile nodes. Fig. 1 shows an example of an ad hoc network, where there are numerous combinations of transmission areas for different nodes. From the source node to the destination node, there can be different paths of connection at a given point of time. But each node usually has a limited area of transmission as shown in Fig. 1 by the oval circle around each node. A source can only transmit data to node B but B can transmit data either to C or D. It is a challenging task to choose a really good route to establish the connection between a source and a destination so that they can roam around and transmit robust communication. There are four major ad hoc routing protocols AODV, DSDV, DSR, TORA. All these protocols are constantly being improved by IETF [1]. As a result, a comprehensive performance evaluation is of ad hoc routing protocols essential. In this work, OPNET simulator version is used to simulate three ad hoc routing protocols, that is, DSDV, AODV, and TORA. We evaluated all available metrics and then performed a comparative performance evaluation. Since these protocols have different characteristics, the comparison of all performance differentials is not always possible. The following system parameters are taken for the simulation of all the above scenario at channel speed 11Mbps and simulation time 20 minutes. The comparative studies of the simulation results for these parameters are also reported.

(i) Number of hops per route,

(ii) Traffic received and sent,

(iii) Total route requests sent,

(iv) Total route replies sent,

(v) Control traffic received and sent,

(vi) Data traffic received and sent,

(vii) Retransmission attempts,

(viii) Throughput,

To the best of our knowledge, very few papers have been published. In section 2, we review the mostly used wireless ad hoc protocols. In Section 3, we present the performance metrics of our simulation. Section 4 performance comparison of the protocols. We draw our conclusions in Section 5.

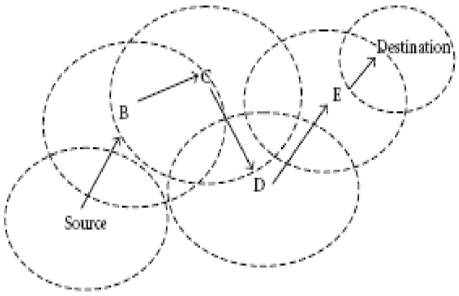

Fig. 1 Ad hoc networking model

II. AD HOC ROUTING PROTOCOLS

Among the various ad hoc routing protocols proposed in the literature [1, 2], TORA, DSR, and AODV appear to be the most promising. TORA [3, 4] is a distributed routing protocol for ad hoc networks, TORA is designed to minimize reaction to topological changes. A key concept in its design is that control messages are typically localized to a very small set of nodes. It guarantees that all routes are loop-free (temporary loops may form), and typically provides multiple routes or any source/destination pair. It provides only the routing mechanism and depends on Internet MANET Encapsulation Protocol (IMEP) [5, 6] for other underlying functions. This uses a link reversal algorithm.. TORA involves four major functions: creating, maintaining, erasing, and optimizing routes [7–9].

The DSR protocol [1, 10, 11] also belongs to the class of reactive protocols and allows nodes to dynamically discover a route across multiple network hops to any destination. Source routing means that each packet in its header carries the complete ordered list of nodes through which the packet must pass. DSR uses no periodic routing messages (e.g. no router advertisement), thereby reducing network bandwidth overhead, conserving battery power and avoiding large routing updates throughout the ad hoc network. Instead DSR relies on support from the MAC layer (the MAC layer should inform the routing protocol about link failures). The two basic modes of operation in DSR are route discovery and route maintenance [9].

The AODV algorithm [12] is based upon the distance vector algorithm (i.e. AODV only requests a route when neded and does not require nodes to maintain routes to destinations that are not actively used in communications). It shares on-demand characteristics of DSR, and adds the hop-by-hop routing, sequence numbers, and periodic beacons from DSDV[13]. It has the ability to quickly adapt to dynamic link conditions with low processing and memory overhead. AODV offers low network utilization and uses destination sequence number to ensure loop freedom. It is a reactive protocol implying that it requests a route when needed and it does not maintain routes for those nodes that do not actively participate in a communication. An important feature of AODV is that it uses a destination sequence number, which corresponds to a destination node that was requested by a routing sender node. The destination itself provides the number along with the route it has to take to reach from the request sender node up to the destination. If there are multiple routes from a request sender to a destination, the sender takes the route with a higher sequence number. This ensures that the ad hoc network protocol remains loop-free. AODV keeps the following information with each route table entry [12]:

(i) destination IP address ,

(ii) destination sequence number,

(v) hop count, that is, number of hops required to reach the destination,

(vi) next hop (i.e. the neighbor, which has been designated to forward packets to the destination for this route entry)

(vii) request buffer,

(viii) lifetime, (i.e. the time for which the route is considered valid)

III. PERFORMANCE METRICS

We evaluated key performance metrics for three different applications using DSR, TORA, and AODV protocols. The parameters used for wireless LAN application performance evaluation include: control traffic received and sent, data traffic received and sent, throughput, and retransmission attempts.

We used the following parameters for evaluating the effect of variation on different protocols: routing traffic received and sent, total traffic received and sent, number of hops, route discovery time, and ULP traffic received and sent, throughput.

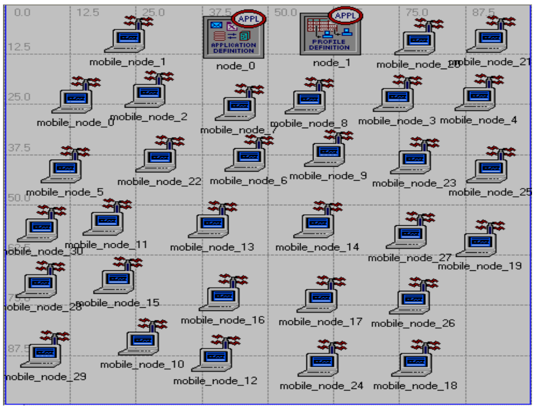

Fig. 2 A proposed model of the ad hoc network.

IV. PERFORMNACE COMPARISON OF THE PROTOCOLS

For all simulations, the same movement models were used, and the number of traffic sources was fixed at 30. Figure 2 shows a model of nodes used to simulate different ad hoc network protocols. A square of 20 meters is used to define the area of node's mobility. In the simulation, the following parameters are used:

(i) duration: 20 minutes,

(ii) speed: 256, 512, 1024

(iii) nodes: 30

A. *Wireless LAN*

Figure 3 shows the control traffic received in packets/s for DSR, TORA, and AODV protocols for a wireless LAN application. Figure 2 shows that the TORA protocol performs better than the other two. Although AODV does not perform well at the beginning, later it does well. DSR's performance remains average during the entire evaluation time. Figure 4 shows the control traffic sent in packets/sec. It is obvious that

TORA performs better than AODV and DSR. Although DSR and AODV have shown an average performance throughout the entire simulation, they show better performance compared to TORA at the end. TORA uses a fast router-finder algorithm, which is critical for TORA's better performance.

Both DSR and AODV have to go through route creation using RREQ and RREP messages. Once the routes are created, DSR and AODV tend to do better than TORA. As a result, we observe from Figures 3 and 4 that, near the end of simulation time, both AODV and DSR show better performance than TORA.

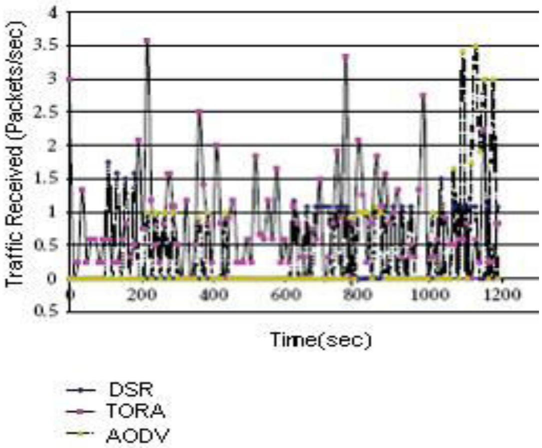

Fig. 3 Control traffic received for different protocols in wireless LAN.

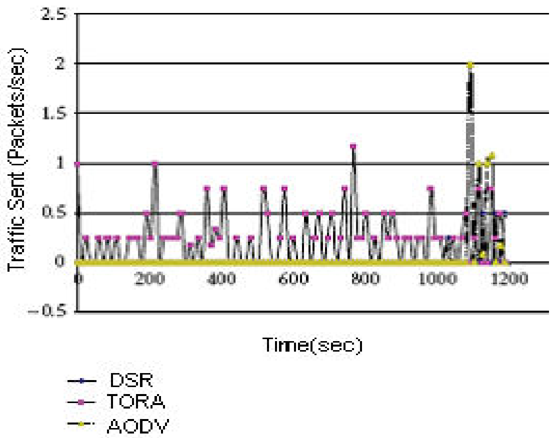

Fig. 4 Control traffic sent for different protocols in wireless LAN

Figures 5 and 6 show the data traffic received and data traffic sent in packets/sec, respectively, for DSR, AODV, and TORA protocols. From Figure 5, it is evident that, at the beginning of the simulation TORA appears to dominate over AODV and DSR, but at the end, AODV yields the best result. DSR shows poor performance and the traffic remains always at the lower level, whereas AODV performs well most of the time. In Figure 6, we observe that

TORA performs well during most of the simulation time. AODV shows consistent performance and peaks at the end of the simulation. DSR does not show any positive traffic except for the last few seconds of the simulation. Figure 7 shows the throughput in bits/sec for DSR, TORA, and AODV protocols, where AODV shows significantly better performance than the other two, and TORA performs slightly better than DSR. Figure 8 shows the retransmission attempts in packets/sec as a function of time for wireless LAN involving different protocols. It is evident from Figure 8 that TORA requires a lot of retransmission attempts before it can successfully transmit data due to the fact that only TORA uses UPD packet. When a node first gets a QRY message for a destination, if it does not have a route for the requested destination, it broadcasts a UPD message and increases the height of the node. In this way, it tries to transmit the UPD message until it gets the destination node. DSR and AODV have almost the same logic to find a route and show almost similar performance near the end of the simulation time.

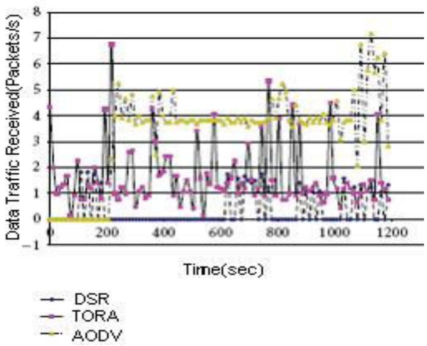

Fig. 5 Data traffic received for different protocols in wireless LAN

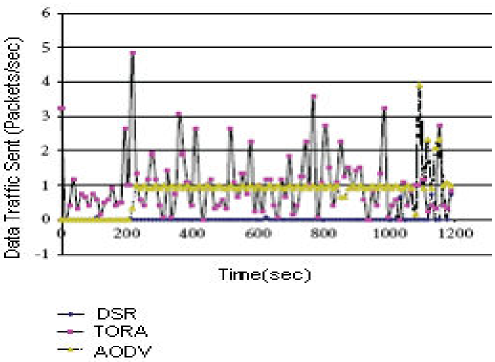

Fig. 6 Data traffic sent for different protocols in wireless LAN

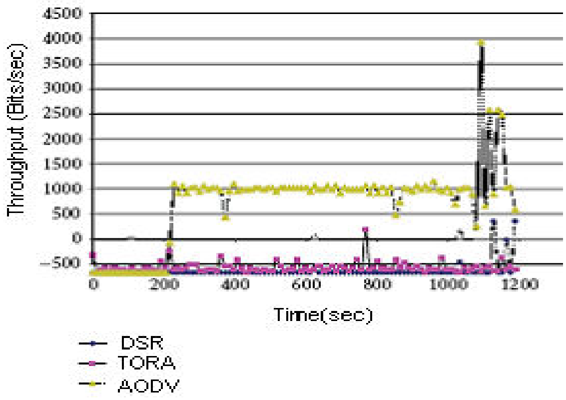

Fig. 7: Throughput of different protocols in wireless LAN

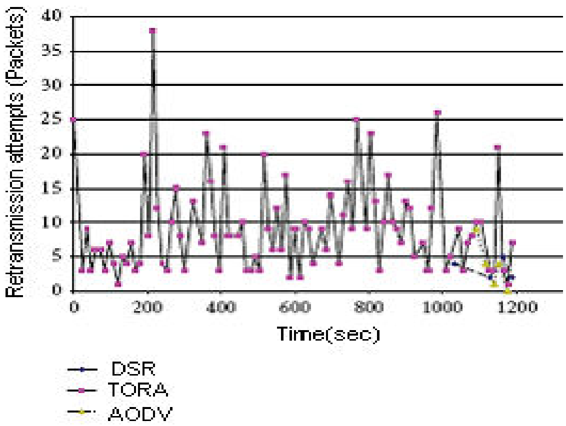

Fig. 8 Retransmission attempts for different protocols in wireless LAN

V. CONCLUSION

In this paper, OPNET Simulator has been used, we evaluated the performance of widely used ad hoc network routing protocols. The simulation characteristics used in this research, that is, the control traffic received and sent, data traffic received, throughput, retransmission attempts, and traffic received, are unique in nature, and are very important for performance evaluation of any networking protocol.

Performance evaluation results for some ad hoc network protocols were previously reported [1, 14], which primarily covered the impact of the fraction of packets delivered, end-to-end delay, routing load, successful packet delivery, and control packets overhead. In this paper, we perform a thorough analysis that includes additional parameters. For comparative performance analysis, we first simulated each protocol for ad hoc networks with 30 nodes. In case of wireless LAN, TORA shows good performance for the control traffic received, control traffic sent, and data traffic sent. However, AODV shows better performance for data traffic received and throughput. DSR and AODV show poor performance as compared to TORA for the control traffic sent and throughput. However, TORA and AODV show an average level of performance for the data traffic received and data traffic sent, respectively.

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EMERGING TECHNOLOGIES FOR INERTIAL SENSORS

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ABSTRACT

In this paper, a review is given of the current developments in the area of inertial sensor technology (Gyroscope and accelerometer) for the military applications in particular and civil application in general. The ongoing research works in development of inertial sensor technology are brought out. In the military aircraft (Multi-role fighter aircraft) micromechanical sensors and improved fibre-optic gyroscopes are replacing many inertial navigation systems (INS) which were using ring laser gyros (RLG) and mechanical accelerometers. The research is driven not only by cost but alsoby environmental factors (Vibrations, temperature etc) in military applications. Inertial Navigation System (INS) integrated with global positioning system (GPS)/GLONOSS (Or future ISRO navigation satellite system) will lead to large volume applications in missile guidance, guided artillery etc.

Keywords: Global Positioning System; Inertial Navigation System; Gyroscopes; accelerometers; Corolis effect; Sagnac Effect.

ABBREVIATION INDEX

MEMS	Micro-electro-mechanical systems	INS	Inertial Navigation system
MVG	Micromachined vibrating gyroscope	GPS	Global positioning system
PVG	Piezoelectric vibrating gyroscope	IMU	Inertial measuring unit
TFG	Tuning Fork Gyroscope	RLG	Ring Laser Gyro
MSG	Magnetically suspended gyroscope	FOG	Fibre optic gyro
MFOGS	Micro fibre optics gyros	MTBF	Mean time between failures
MAGS	Micro atoms gyros	CNT	Carbon Nano Tube

I. INTRODUCTION

Gyroscopes (Abbreviated as Gyros) and accelerometers together are called inertial sensors because they exploit the properties of inertia (Rigidity and Precession) in case of Gyros and changes in linear motion in the case of the accelerometer. Without them an unstable military aircraft cannot be flown or controlled. They determine the performance and accuracy of navigation and guidance system. Driving force behind introducing new technology has been to improve performance (navigation accuracy) and reliability. Another factor which has led to significant development in inertial technologies is external aiding. GPS, Doppler radar used along with Kalman Filter and integrated with INS has drastically improved the performance by overcoming inertial sensor drift. This paper presents the current trends in inertial sensor technologies in today's world.

II. Inertial Sensor Technology trends

Technology developments for new inertial sensors are underway for commercial and military applications. They will have applications in tactical guidance and navigation mission. These are basically silicon MEMS, micro atom, piezoelectric vibrating, Micromachined vibrating and silicon accelerometers.

A. Fibre-Optic Gyros (FOGs). FOGs operate on the principle of 'Sagnac Effect'. They measure the angular rate of rotation by sensing the difference in transit time for light waves travelling in opposite directions. The time difference ΔT is directly proportional to the input rate of rotation. Fig 1 below illustrates the configuration of the FOGs interferometer. Change in frequency Δ f between the two waves is given by

$$\Delta f = K_0 \, \dot{\theta} = K_0 \, \omega$$

FOG is implemented using a fibre-optic sensing coil, an integrated optics chip, and a light source and photo diode detector. Various elements are integrated onto a substrate which leads to low manufacturing cost.

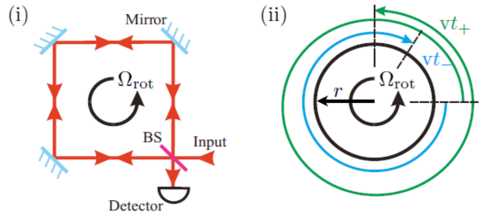

Fig 1: Sagnac Effect

IFOG offers many advantages in terms of reliability, ruggedness and performance. Integrated 3-axis FOGs are produced all of them sharing the same laser source. However in order to achieve inertial quality performance, effect of environmental noise must be minimized. Otherwise these effects will swamp the small Sagnac phase shifts being measured. A Honeywell made IFOG has been reported to have bias stability of less than 0.0003° / hour and scale factor stability of less than 1 parts per million. 90% of the present market share in aviation is reported to be dominated by RLG because of its reliability. However it involves high quality machining operations, high quality mirrors and high cost. Against this, FOG does not require very precision machining and they also do not exhibit lock-in at low rates. FOGs are finding application in sensing automobile industry in sensing vehicle skid (Yaw rate) and other commercial applications at much lower cost than RLG.

B. Micro mechanical Gyros (or MEMS Gyros). These gyros are based on the effect of Coriolis forces which are experienced when a vibrating mass is subjected to a rate of rotation about an axis which is in the plane of vibration.

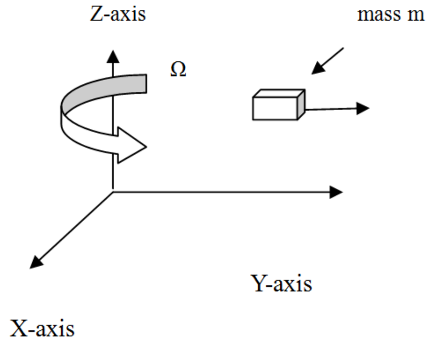

Fig 2: Coriolis Acceleration

Coriolis acceleration is given by the equation:

$$\overrightarrow{a_{cor}} = 2\,\overrightarrow{V_{pm}} \times \vec{\Omega}$$

Where; $\overrightarrow{a_{cor}}$ = Coriolis acceleration;

$\overrightarrow{V_{pm}}$ = Velocity of the proof mass; $\vec{\Omega}$ = Rate of rotation.

Micro machined sensors are being used not only in aero space such as flight control systems, missile guidance. They also find application in car stability system, robotics, antenna stabilization system, automatic vehicle control, camera stabilization system. They use micro-machining technologies and integrated circuit manufacturing technologies. They provide advantages of being robust, no wearing parts and very high mean time between failures (MTBF) in excess of 100,000 hours. A MEMs sensor is shown in Fig 3 & 4 [Ref 1].

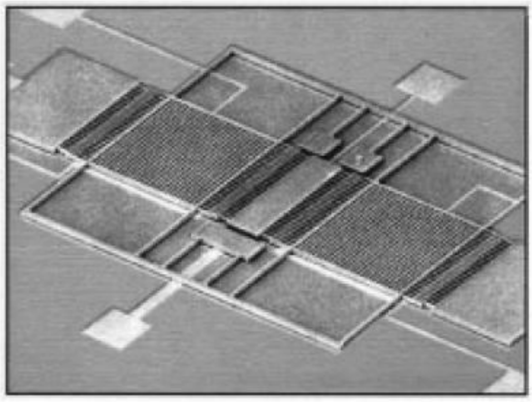

Fig 3: Micro mechanical tuning fork Gyro

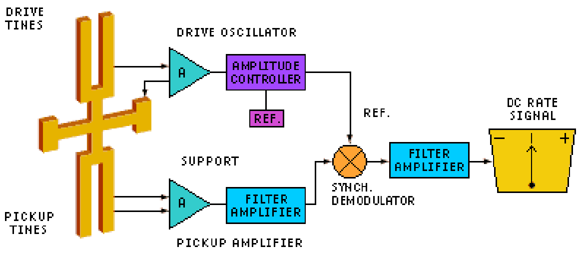

Fig 4 Quartz rate sensor

Though tuning fork gyros were invented as back as 1940 by F W Meredith in 1942, it is micro-machining and semi-conductor technology which has enabled the advanced technologies.

C. Nano-Gyroscope. Nano-scale gyroscopes with vibratory carbon nano-tubes (CNT) consume less energy have high resolution. A CNT gyroscope based on the Coriolis Effect is shown in Fig 5. As seen in the fig, CNT is vibrated in the x-direction where rotation is applied about the Z-direction; Coriolis force is sensed along the Y-direction. The carbon nano- tube is forced to vibrate to its natural frequency by the electrostatic force. An electrode is set near the tip of CNT tip, which pulls the CNT for mechanical resonance by the applied Alternating Current voltage. The CNT gyroscopes can give sensitivity up to 100 rad/s with a resonant frequency of 1 MHz

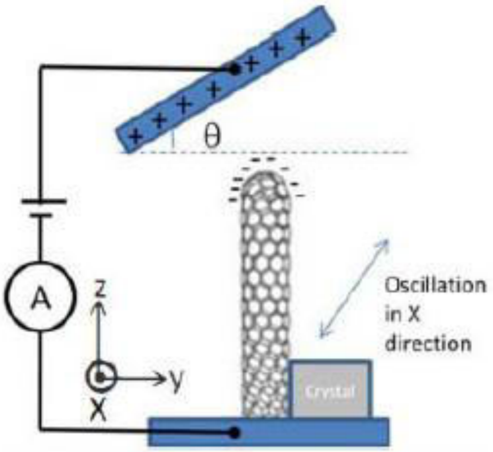

Fig 5: CNT Gyroscope

D. Micromechanical Accelerometers. The measurement of vehicle acceleration is based on Newton's second law of motion: **Force = mass × acceleration**. However it is impossible to differentiate between the force acting on a suspended mass due to gravity and force required to accelerate the mass. Hence an accelerometer measures the specific force component along its input axis and not the vehicle acceleration component. Mechanical accelerometers are simple spring restrained pendulous type or closed loop torque balance type. Fig 6 is schematic of a closed loop torque balance accelerometer. Bandwidth of these accelerometers is approximately 500 Hz. Temperature compensation is generally required for temperature dependent scale factor. Size is about 1 inch diameter and 1 inch length. Micro mechanical sensors fabricated from silicon using semi-conductor technology are being used where accuracy requirements are not very high.

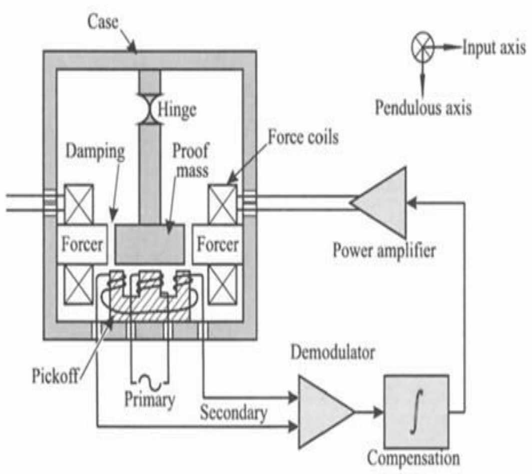

Fig 6: Torque balance pendulous accelerometer

They are low cost and small in size. Micro mechanical accelerometers are pendulous type and use closed loop capacitive sensing and electro static torquer. Fig 7 is an example of a force balance micro mechanical accelerometer. It is a monolithic silicon structure consisting of a pendulum and electrostatic torque. Size of such sensors is typically 300 × 600 μ m. Torque required to rebalance the plate is proportional to the input acceleration.

Performance around 100 μg bias errors has been reported to be achieved. Further improvements are expected to meet the accuracy requirement for strap down INS.

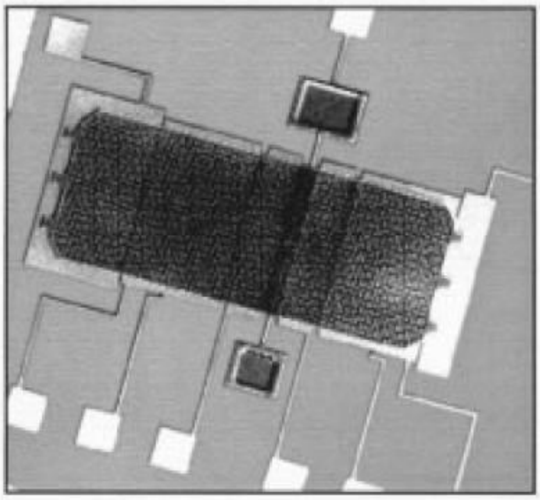

Fig7: Micro mechanical pendulous rebalance accelerometer

III. RESULTS & DISCUSSION

As technology improves, solid state gyroscope and accelerometers will see wide applications in the field of defense and consumer electronics. The Military application requires gyroscopes and accelerometers with very high performances, lower volume/weight and most importantly environmental tolerances (Temperature, shock, vibrations and humidity). Accuracy requirements for flight control systems and strap down INS system is given in table 1 below. Gyros operating on 'solid state' principles have been developed because of higher reliability and lower cost. At the lower end of the accuracy spectrum, micro machined vibrating mass rate gyros exploiting semi-conductor technology have become well established. They are being used as flight control system and solid state standby artificial horizon. They offer major advantage of extreme reliability, low power consumption, robustness and relatively low cost.

Table 1: Accuracy Requirements

		Flight Control System	Strap Down INS
Gyro	Scale factor	0.5%	0.001% 10 ppm
	Zero offset/rate uncertainty	1° /min	0.01° /hour
Acctr	Scale factor	0.5%	0.01% 100 ppm
	Zero offset/rate uncertainty	5×10^{-3} g	5×10^{-5} g

Among micro- machined vibrating gyroscopes, tuning fork gyroscopes and vibrating ring gyroscopes have demonstrated good performance. Bias drift of about 0.1° /hour has been achieved. Performance however is severally affected because of vibration, poor robustness to shock etc. To be used in inertial measuring unit (IMU), gyroscopes need to be integrated with accelerometers. Integration will lead to large volume on a single chip. This is being solved by using multi-axis gyroscope. Piezoelectric vibrating gyroscopes are simple, robust, have large measuring range. They also have good resistance to shocks. They can work in adverse environment and require no special vacuum packaging. Other area of development is magnetically suspended gyroscope (MSG). They have no supporting mechanical structure, are insensitive to stresses & temperature variations. Also they are capable of measuring multi-angular rates & accelerations which lead to size reduction when integrated with inertial measuring unit (IMU) Limitation of the MSG is that it requires a permanent magnet to suspend the rotor and hence the volume is large to integrate on a single chip.

IV CONCLUSION

The technology of inertial sensors is moving with a very rapid pace. In future, it is projected that INS system will have navigation accuracy as good as GPS and one would be able to navigate with smart phone using only INS.

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SIGNIFICANCE OF SAMPLING RATE CONVERTERS IN WSNS

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***Abstract*— A wireless sensor network (WSN) is a group of specialized transducers with a communications infrastructure for monitoring and recording conditions at diverse locations. A sensor network consists of multiple detection stations called sensor nodes and sampling rate offsets exist between these nodes. As nodes utilize individual clock sources, sampling rate offsets are inevitable and may cause severe performance degradation. Hence Sampling Rate converter (SRC) becomes very important and integral part of the multi-rate wireless sensor network.**

Keywords—Wireless Sensor Networks, sensor nodes, sampling rate offsets, SRC.

I. INTRODUCTION

Sample rate conversion, or interpolation and decimation as they are known, are a clever digital signal processing techniques that broadband and wireless design engineers can employ during the system design process. Using these techniques, design engineers can gain an added degree of freedom that could improve the overall performance of system architecture.

Re-sampling is usually done to interface two systems which have different sampling rates. If the ratio of two system's rates happens to be an integer, decimation or interpolation can be used to change the sampling rate (depending on whether the rate is being decreased or increased); otherwise, interpolation and decimation must be used together to change the rate.

Crochiere and Rabiner [1] present a good background for the sampling-rate conversion problem, and describe the “classical” rational-ratio design of implementing the ratio L/M as an up-sampling by a factor of L followed by appropriate filtering, followed by a down-sampling by a factor of M. This method tends to be useful for small values of L and M; otherwise the intermediate sampling-rate tends to get unwieldy.

A practical and well-known example results from the fact that professional audio equipment uses a sampling rate of 48 kHz, but consumer audio equipment uses a rate of 44.1 kHz. Therefore, to transfer music from a professional recording to a CD[2], the sampling rate must be changed by a factor of: (44100 / 48000) = (441 / 480) = (147 / 160)

There are no common factors in 147 and 160, so we must stop factoring at that point. Therefore, in this example, we would interpolate by a factor of 147 then decimate by a factor of 160.

II. METHODS OF SAMPLING RATE *CONVERSION*

Interpolation

"Up-sampling" is the process of inserting zero-valued samples between original samples to increase the sampling rate. (This is called "zero-stuffing".) Up-sampling adds to the original signal undesired spectral images which are centred on multiples of the original sampling rate.

$$y(n) = ([\uparrow L]\, x(n)) * h(n) = \sum_k x(k)\, h(n - L\, k)$$

"Interpolation", is the process of up-sampling followed by filtering. In order to remove the undesired spectral images, the following system is used for interpolation.

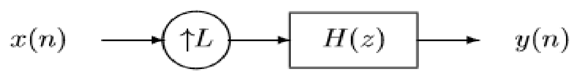

Figure1. Interpolation

The combined up-sampling and filtering can be written as

The filter fills zero’s that are introduced by the up-sampler. Equivalently, it is designed to remove the spectral images. It should be a low-pass filter with a cut-off frequency $\omega_o = \pi/L$. In this context, the low-pass filter is often called an interpolation filter.

Decimation

The following system is used for decimation.

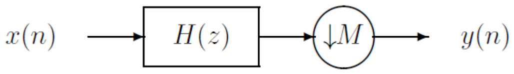

Figure 2. Decimation

The combined filtering and down-sampling can be written as

$$y(n) = [\downarrow M]\,(x(n) * h(n)) = \sum_k x(k)\,h(M\,n - k).$$

The filter is designed to avoid aliasing. It should be a low-pass filter with a cut-off frequency $\omega_o = \pi/M$. In this context, the low-pass filter is often called an anti-aliasing filter.

Fractional Sampling rate conversion

A rate changer for a fractional change (like 2/3) can be obtained by cascading an interpolation system with a decimation system. Then, instead of implementing two separate filters in cascade, one can implement a single filter. Structure for rational rate changer:

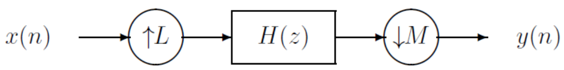

Figure 3. Fractional SRC

The filter is designed to both eliminate spectral images and to avoid aliasing. The cascade of two ideal low-pass filters is again a low-pass filter with a cut-off frequency that is the minimum of the two cut-off frequencies. So, in this case, the cut-off frequency should be

$$\omega_o = \min\left\{\frac{\pi}{L}, \frac{\pi}{M}\right\}$$

When the ratio between the desired sample rate and the actual symbol rate is an integer, the up-sampling or down sampling process is straight forward. However, there are many applications where the amount by which the discrete time signal must be up-sampled or down-sampled is not always fixed at an integer. In this case, an SRC method capable of handling arbitrary conversion ratios is required.

These all constraints can be answered by the design of the filter H(z).Hence the major challenge tends to be the implementation of the filter H(z) and/or how they update the filter coefficients in order to be as efficient as possible.

Performance Limits of Sampling rate Converter

The performance of the sampling-rate converter algorithm is determined by 5 design parameters:

1. The length L of the sub filters (or equivalently, the length of the prototype low pass filter)
2. The technique (Parks-McClellan, Kaiser,etc) used to design the prototype low pass filter.
3. Allowable pass band and stop band ripple in the prototype low pass filter.
4. The number of sub filters m
5. The order of the polynomial interpolation.

III NEED OF SRC IN WIRELESS SENSOR NETWORKS

Wireless sensor networks (WSNs) have gained world-wide consideration in recent years, particularly with the proliferation in Micro-Electro-Mechanical systems (MEMS) technology which has facilitated the development of intelligent sensors. These sensors are small, with limited processing and computing resources. These sensor nodes can sense, measure, and collect information from the environment and, based on some local decision process, they can transmit the sense data to the users. The sensors nodes consist of sensing, data processing, and communicating component, leverage the idea of sensors networks. The components of Sensor nodes are as shown in the figure.

A sensors network [3] is composed of a large number of sensor nodes that are densely deployed either inside the

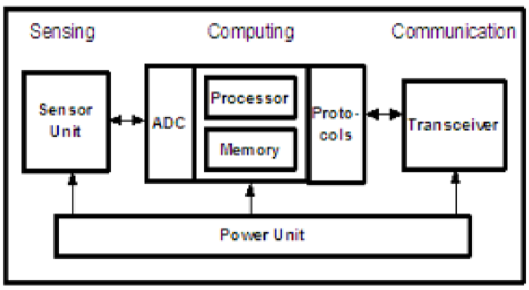

Figure 4. Components of Sensor nodes

phenomenon or very close to it. Intelligent sensor nodes are low power devices equipped with one or more sensors, a processor, memory, a power supply, a radio and an actuator. Experimental measurements have shown that generally data transmission is very costly in terms of energy consumption, while data processing consume significantly less [4]. Communication subsystem has energy consumption much higher than the computation subsystem. The energy expenditure of transmitting a single bit of information is approximately the same as that needed for processing a thousand operations in a typical sensor node [5]. Therefore Communication should be traded for computation in WSNs.

For example consider a variety of mechanical, thermal, biological, chemical, optical, and magnetic sensors attached to the sensor node to measure properties of the atmosphere. Since the sensor nodes have limited memory and are typically deployed in difficult-to-access locations, a radio is implemented for wireless communication to transfer the data to the base station.

But as nodes utilize individual clock sources, sampling rate offsets are inevitable and may cause severe performance degradation.

For this purpose wireless transceivers are required that need to support various sampling rates in either a receive chain or transmit chain to accommodate different system bandwidths, or different operating bands used in such transceivers. One part of accommodating different bandwidths, such as in baseband processing of a transceiver, is through sampling rate conversion that converts one sampling rate of a signal to another sampling rate. Such conversion could be performed at the output of an analog-to-digital converter (ADC), the input of a digital-to-analog converter (DAC), or any other portions of baseband processing utilizing sampling of signals requiring conversion or adjustment of the sampling rates. Conventional sampling rate conversion (also termed herein as “re-sampling”) used in transceivers to accommodate different bandwidths or bands may include integer down-sampling (i.e., decreasing the rate at which a signal is sampled by an integer factor) or up-sampling (i.e., increasing a sampling rate of a signal by an integer factor) or fractional sampling (i.e., changing the sampling rate according to a predetermined fractional value).

As in wireless sensor networks low power consumption is of utmost importance, our concentration can be on efficient structure of SRC which in turn can also influence the power consumption in communication subsystem of sensor node.

IV STUDY OF TWO SAMPLING RATE CONVERTERS

We analyzed the new re-sampling algorithm proposed by Laurent de Soras, in his paper “The Quest for the Perfect Re-sampler” [6] against the most common poly phase sampling rate conversion method. We implemented a fixed point variant of the proposed algorithm with further enhancements to meet the memory and cycle requirements without compromising on the quality.

We used an open source code SoX Re-sampler library as reference for the Laurent de Soras – re-sampler implementation and compared its performance and quality against the benchmarks set by Qualcomm’s re-sampler implementation which is based on poly phase sampling rate converter. [For comparison sake in theory and study we used an open source poly phase sampling rate converter – ScopeFIR / libresample]

Comparison of libresample vs SoX library [7] for sine sweep is shown in Figure 5. and sine tone is shown in Figure 6. SNR and aliasing effect in this case shows SoX resampler’s superior quality.

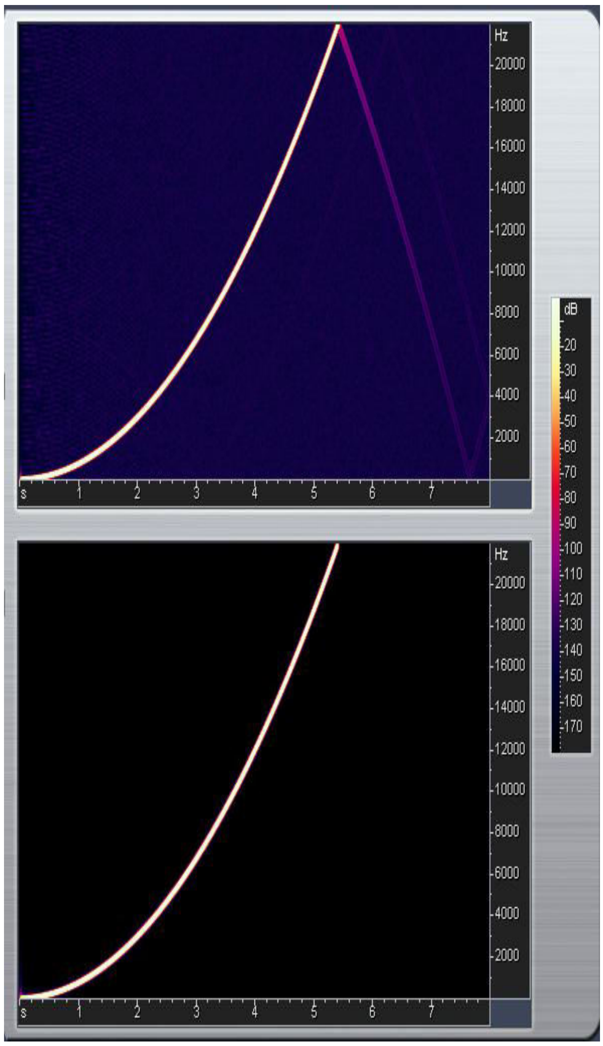

Figure 5. Comparison of Libresample [top] vs SoX [bottom] library for sine sweep-downsampling

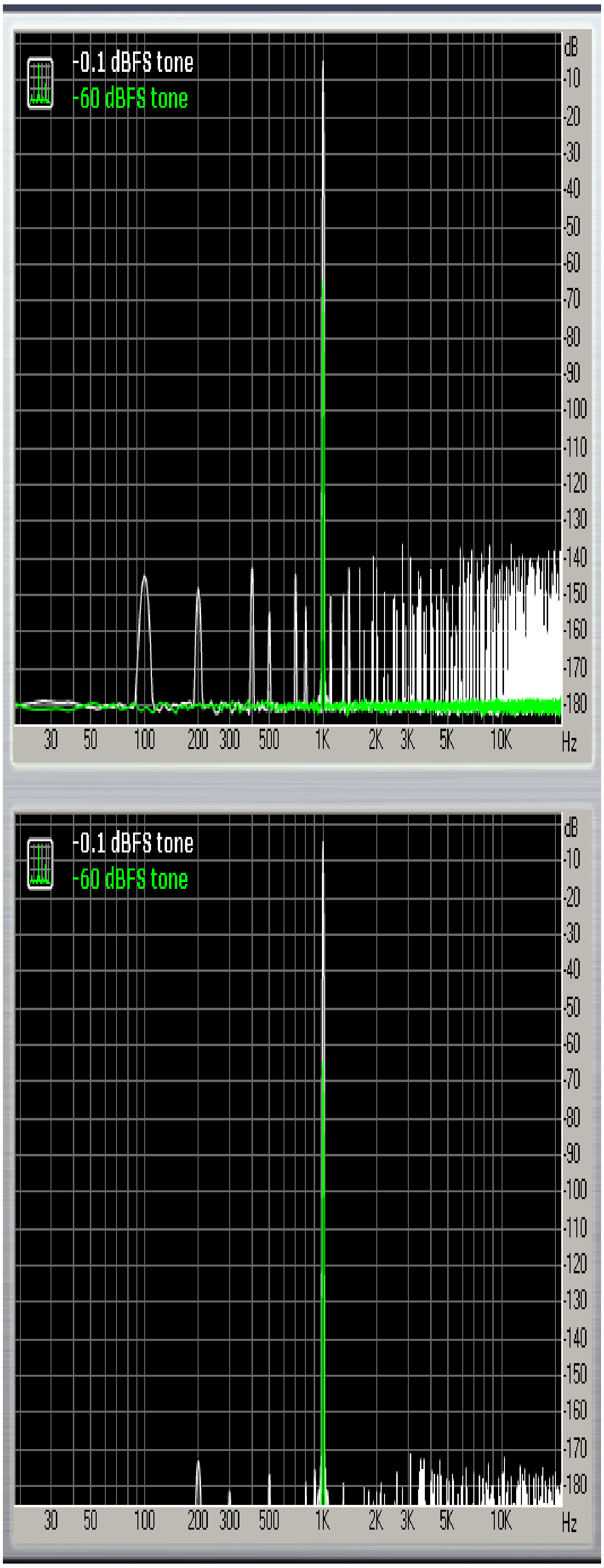

Figure 6. Comparisonof Libresample[top] and SoXresampler [bottom] for 1KHz Sine Sweep.

V. THEORY AND RESULTS FOR THE FIXEDPOINT RESAMPLER

The implementation below provides signal evaluation at an arbitrary time, where time is specified as an unsigned binary fixed-point number in units of the input sampling (assumed constant).

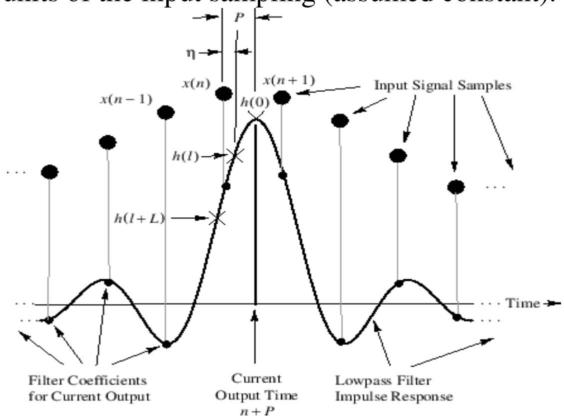

Figure 7. Sinc Interpolation

In a fixed point re-sampler the time register is divided into three fields: The leftmost field gives the number *n* of samples into the input signal buffer, the middle field is an initial index *l* into the filter coefficient table *h(l)*, and the rightmost field is interpreted as a number *n* between 0 and 1 for doing linear interpolation between samples *l* and *l+1* (initially) of the filter table.

The concatenation of l and n are called *P= (0,1]* which is interpreted as the position of the current time between samples *n* and *n+1* of the input signal.

Let the three fields have n_n, n_l, and n_{nn} bits, respectively. Then the input signal buffer contains $N=2^n{}_n$ samples, and the filter table contains $L=2^n{}_l$ ``samples per zero-crossing." (The term ``zero-crossing" is precise only for the case of the ideal low pass; to cover practical cases we generalize ``zero-crossing" to mean a multiple of time $t_c=0.5/f_c$, where f_c is the low pass cutoff frequency in cycles per sample.) For example, to use the ideal low pass filter, the table would contain *h(l)=Sinc(l/L)*.

The existing reference code from soxr library has the following characteristics

Worst case Signal-to-Noise Ratio : 145.68 dB

Measured -3dB roll off point : 96.08 %.

Half length of sinc function : 340238 (Float values)

The fixed point re-sampler based on soxr library for the embedded systems has shorter filter sinc values saving memory. The internal main filter kernel has been optimized to fixed-point arithmetic for faster computation time on DSP or ARM based processors. The quality metrics obtained for the fixed point implementation are as below.

Worst case Signal-to-Noise Ratio: 100.43 dB

Measured -3dB roll off point: 80.23 %.

Half length of sinc function : 2464 (q31 fixed point format)

Frequency Response for sine sweep tone to 19kHz sampled at 44100Hz upsampled to 48kHz – for the reference soxr library and the proposed optimized version of soxr library.

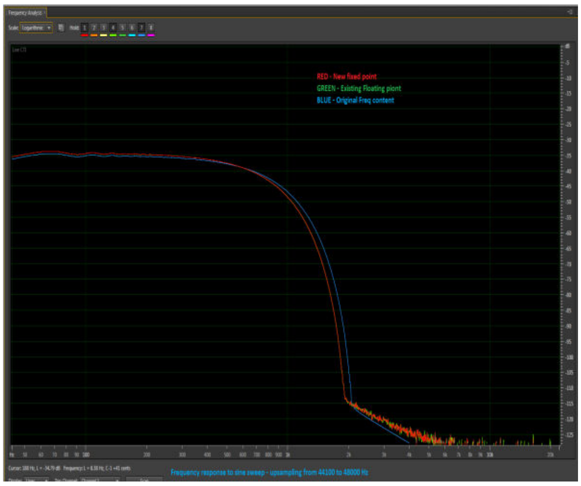

Figure 8. Frequency Comparison of sine sweep input

Frequency Response for sine tone of 997Hz sampled at 44100Hz upsampled to 48kHz – for the reference soxr library and the proposed optimized version of soxr library

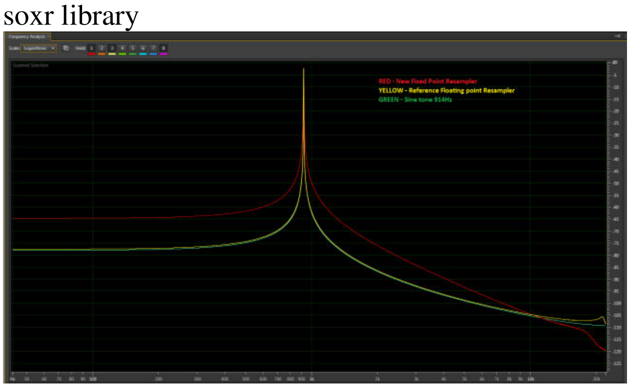

Figure 9. Frequency comparison of 914Hz

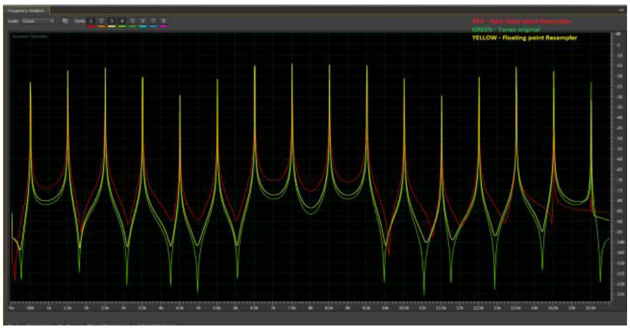

Figure 10. Frequency comparison for multiple tones

Frequency response to multiple tone inputs up sampled from 32000 to 48000Hz

VI CONCLUSION

In this paper we have reviewed the concept of Sampling Rate Converter and its significance in Wireless Sensor Network. It has been observed that a need exists for more efficient and flexible sampling rate conversion for arbitrary sample rates but with less cost in terms of hardware and power consumption. As a case study comparison of two sampling rate converters has been included followed by simulation results of a fixed point re-sampler, where it is noticed that sinc memory length is efficiently reduced, though due to higher noise floor it doesn't meet the industry standard of 110dB SNR. More tweaking and optimizations would be needed to make it viable solution.

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AREA OPTIMIZED AND FREQUENCY EFFICIENT 1024 POINT RADIX-2 FFT PROCESSOR ON FPGA

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Abstract: **This paper presents optimized area and frequency efficient Fast Fourier transform (FFT) processor using radix 2 decimation in time (DIT) algorithm. The proposed FFT processor is a complex FFT processor where a time-multiplexed approach to the butterfly of 1024 point, fixed, 32-bit, based on field programmable gate array (FPGA) is designed. The architecture is based on burst I/O and the pipelined- streaming I/O structure in the butterfly module and the ping-pong operation which is clocking at 480MHz on Xilinx vertex 6 xc6vlx550t-2ff1759.**

Keywords*—Fast Fourier Transform; *Time-Multiplexed Butterfly; Pipelined-Streaming I/O; ping-pong operation; field programmable gate array;

I. INTRODUCTION

The fast Fourier transform (FFT) is one of the most important algorithms in signal processing. Many hardware FFT architectures have been proposed with the aims of speeding up the calculation of the FFT and reducing the amount of hardware resources. Computing the Discrete Fourier Transform (DFT) [1] of 'N' points in the naive way takes O(N2) arithmetical operations. To calculate DFT with reduced number of arithmetical operations from O(N2) to N/2(log2N) FFT algorithm [2] is performed, by taking full advantage of Symmetry and Periodicity of Twiddle Factors. Cooley and Tukey [3] proposed the fast Fourier transform (FFT) algorithm, their work was first published in 1965.

In the realization of FFT, the most generally hardware realization methods are included of DSP (Digital Signal Processor), FFT dedicated chip and FPGA [4].General purpose DSP programmable chips are used for the implementation of DSP algorithms, for high performance applications, special purpose fixed function DSP chipsets or FPGAs are used.

The FPGA is suitable for the high-speed signal processing system owing to its parallel signal processing architecture. FPGA is suitable for the FFT algorithm and has superiorities in performance, costing and power consumption. It is realized by the related EDA (Electronic Design Automation) software and hardware description language. In modern signal processing, the requirements of high speed and reliability are becoming a hot research point. Furthermore, many researchers are researching in combining the real-time requirement of FFT with the flexibility design by the FPGA, realizing the optimal configuration of the parallel algorithm and hardware structure, and improving the FFT processing speed.

According to the requirements, this paper presents a method which realizes the FFT operation based on the FPGA [5]. The FFT design coded in Verilog HDL [6], The FFT design is synthesized on the Xilinx ISE 14.7.

II. COOLEY-TUKEY FFT ALGORITHM

The Cooley-Tukey FFT algorithm [3] provides a systematic solution with a moderate computational complexity. It is based on the observation that multiple operations can be shared when calculating the output frequencies of the FFT. This is done by decomposing the equation of the DFT. The most common decompositions are decimation in time (DIT) and decimation in frequency (DIF).

The DIT decomposition separates the sequence x[n] into its even and odd samples, whereas the DIF decomposition is applied on the output sequence X[k].

$$X[k] = \sum_{n=0}^{N-1} x[n]\, W_N^k \text{ , k= 0,1,…..,N-1}$$

$$\text{Where } W_N^k = e^{-j\frac{2\pi}{N}nk}$$

The odd part and even part of x(n) using radix-2 DIT in (l), obtain the following equations can be obtained

Even terms Odd terms

$$X(k) = \sum_{n=0}^{N/2-1} (x[2n]\, W_N^{2nk}) + \sum_{n=0}^{N/2-1} (x[2n+1]\, W_N^{(2n+1)k}) \forall N \in \mathbb{Z}$$

Where In both decompositions, the N-point DFT is transformed into two N/2-point DFTs. By applying the procedure iteratively, each step halves the number of points of the DFTs, which finally leads to 2-point DFTs. One large computation is reduced to several sequential smaller computations which lead to the radix-2 butterfly as shown

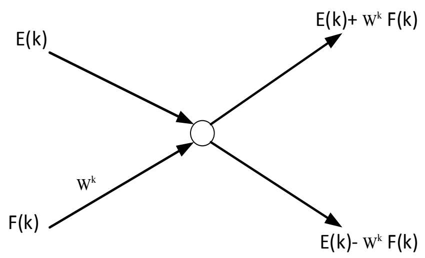

Fig. 1. Basic Butterfly element

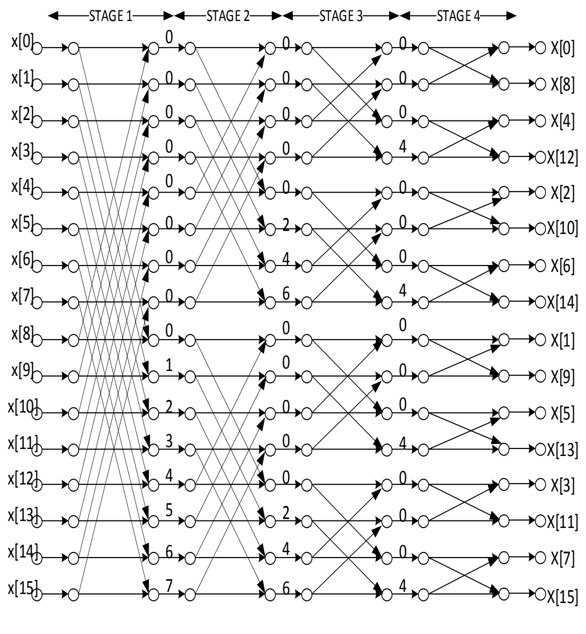

Fig. 2. 16- point radix-2 DIFFFT flow graph.

I. FFT IMPLEMENTATION

In any FFT architecture, for good performance Efficient addressing plays the key role. We efficiently did in place computation in this DIT FFT architecture using dual port RAM's which best suits for radix-2 FFT. The FFT system diagram [1] is shown below.

Fig.3 system diagram shows dual RAM ping pong FFT architecture, where the data reading, processing and writing is done from one RAM to another at each stage and vice versa. To increase the throughput, we are using an extra block RAM named 'Block RAM 3' as shown in Fig.3. Here user no need to wait until the first set data is read and user can read once the output is available while writing the second set of data. In this architecture, we are using pre-computed twiddle factor values.

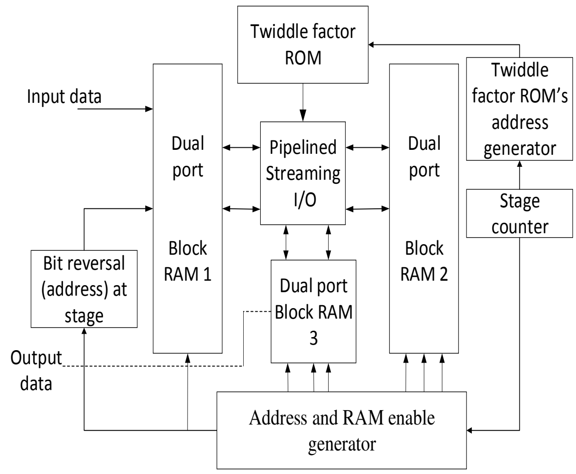

Fig. 3. FFT system diagram

We designed a FFT architecture where we are using pipelined, STREAMING I/O which Allows continuous data processing. Radix-2 butterfly block as show in Fig.4 is 9 stages pipelined as it involves complex multiplications resulting bottle neck for maximum frequency of operation. Where 6 clock cycles for multiplication, 2 clock cycles for addition and 1 clock cycle for signed shift operation.

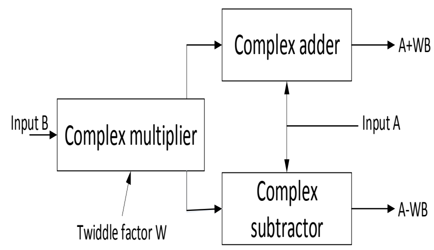

Fig. 4. radix 2 butterfly operation

For one complete FFT operation 512 radix-2 butterflies operation requires total of 10 such stages takes in which 9 clock cycles for pipelining, 8 clock cycles for internal read and write delays. For 1024 radix-2 butterflies operation to load the input data which is nearly 6320 clock cycles for one complete FFT operation.

TABLE I. THREE BITS EXAMPLE OF BIT REVERSAL OPERATION

Index	Address
000	000
100	001
010	010
110	011
001	100
101	101
011	110
111	111

As told in OFT summarizing steps, input should be stored in the reversal address of a one-bit incremental counter. Bit reversal block takes care of reversing input addresses.

PIPELINED, STREAMING I/O

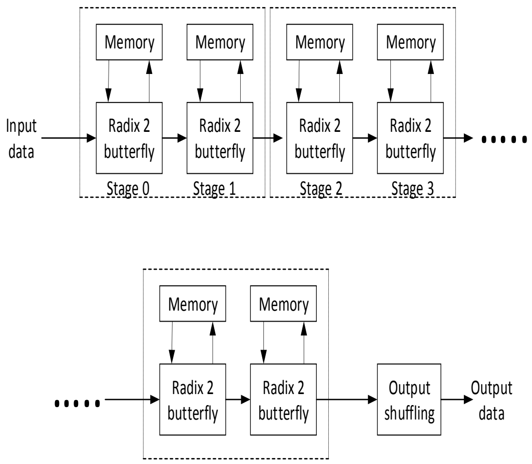

Fig. 5. Pipelined streaming I/O

The Pipelined, Streaming I/O Allows continuous data processing. This solution pipelines several Radix-2 butterfly processing engines to offer continuous data processing. Each processing engine has its own memory banks to store the input and intermediate data. The core has the ability to simultaneously perform transform calculations on the current frame of data, load input data for the next frame of data, and unload the results of the previous frame of data. The user can continuously stream in data and, after the calculation latency, can continuously unload the results.

This architecture covers point sizes from 8 to 65536. The user has flexibility to select the number of stages to use block RAM for data and phase factor storage. The remaining stages use distributed memory as shown in fig 5.

RADIX-2, BURST I/O

The Radix-2, Burst I/O architecture Uses the same iterative approach as Radix-4, but the butterfly is smaller. This means it is smaller in size than the Radix-4 solution, but the transform time is longer. It uses one Radix-2 butterfly processing engine (Figure 6). After a frame of data is loaded, the input data stream must halt until the transform calculation is completed.

Then, the data can be unloaded. As with the Radix-4, Burst I/O architecture, data can be simultaneously loaded and unloaded when the output samples are in bit reversed order.

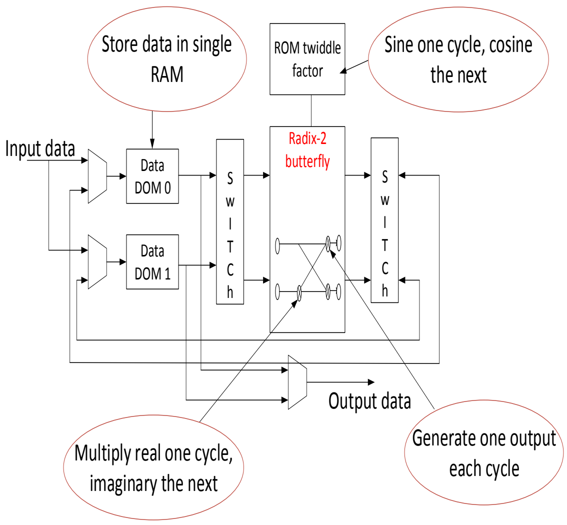

Fig. 6. Radix-2, Burst I/O

This solution sup- ports point sizes from 8 to 65536. Both the data memories and phase factor memories can be in either block RAM or distributed RAM (the latter for point sizes less than or equal to 1024).

ADDRESS GENERATION FOR TWIDDLE FACTOR ROM:

As mentioned, for high throughput and performance of this design efficient addressing is important. To generate addresses to both Twiddle factor ROM and 'In place' computation Block RAM's we used only simple shift operations. The logic to generate twiddle factors addresses to ROM addr_twid, as shown in Fig.6 is as follows in table IV:

TABLE II. LOGIC TO GENERATE TWIDDLE FACTORS ADDRESSES TO ROM

Addresses at	**Stage 1**	**Stage 2**	**Stage 3**	**Stage 4**
n= 2	0	0,1	-	-
n=3	0	0,2	0,1,2,3	-
n=4	0	0,4	0,2,4,6	0,1,2,3 ,4,5,6, 7

As shown in Fig.7 initial shifter value is loaded as N/2 (N umber of twiddle factors for given 'N') with left shifted by one. The one bit counter 'K' is designed such that the logic is as follows:

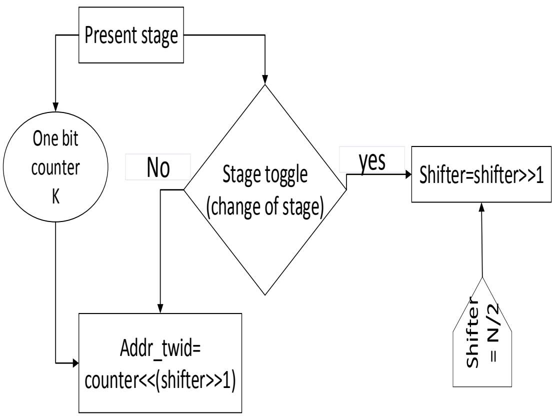

Fig. 7. Flow chart for address generation to the twiddle factor ROM

TABLE III. LOGIC OF ONE BIT COUNTER 'K'

counter value at	**Stage 1 K=**	**Stage2 K=**	**Stage3 K=**	**Stage 4 K=**
n= 2	0,0,0..	0,1,0,1	-	-
n=3	0,0,0	0,1,0,1	0,1,2,3 0,1,2,3	-
n=4	0,0,0	0,1,0,1	0,1,2,3 0,1,2,3	0,1,2,3, 4,5,6,7, 0,1,2,3,

				4,5,6,7

At each stage toggle (i.e. is change of stage) shifter shifts its value left by l(or divided by 2). The shifter values for N = 16 at stage 1, stage 2, stage 3, stage 4 are 8,4,2 and 1 respectively.

ADDRESS GENERATION FOR BLOCK RAM's:

In radix-2 FFT as shown in Fig.4, if the first input A address (Address A) to the Block RAM 1 is 'M' then the input B address (Address B) can be calculated as 'M + $2^n/2$' where 'n' is present stage. The logic to calculate input addresses in block RAM is shown in Table IV.

TABLE IV. LOGIC TO CALCULATE INPUT ADDRESSES IN BLOCK RAM

	M=	**M + 2''/2 =**
n= 1	0,2,4,6''	1,3,5,7''
n=2	0,1,4,5....	2,3,6,7''
n=3	0,1,2,3''	4,5,6,7''

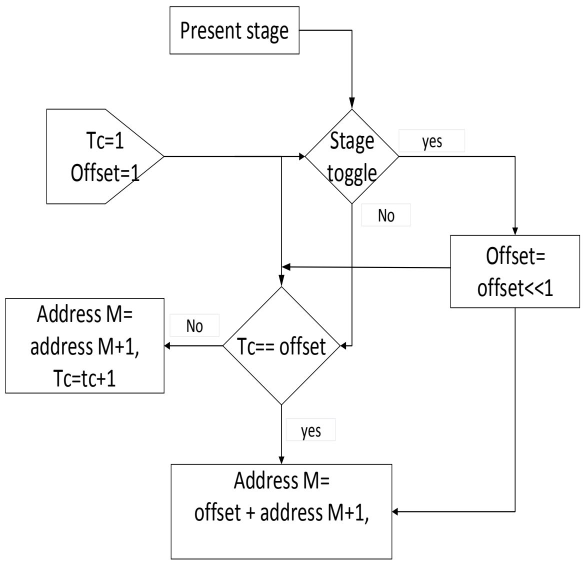

Fig. 8. Flow chart for address generation to the block RAM's

Fig.8 the logic flow to generate value of 'M', where 'tc' in Fig.8 is the terminal counter which resets to '1' if its value equals to the value of offset. It adds offset address value to the present address and resumes the normal counter operation for address and terminal counter.

II. COMPARISION AND ANALYSIS

We implemented this FFT architecture in Xilinx xc6vlx550t-2ff1759 vertex 6 FPGA for length 1024 and the comparison results are shown in table 2, this design is operated on 480 MHz frequency. The proposed work utilizes less area with less number of slice LUT's and flip flop's.

TABLE V. COMPARISON WITH PREVIOUS WORK

	[7]	**Proposed work**
Algorithm	**Radix 2**	**Radix 2**
Chip type	Vertex-6, xc6vlx550t	Vertex-6, xc6vlx550t
Number of points	1024	1024
Operating frequency	385 MHz	480 MHz
Number of slices	2633	2059

TABLE VI. COMPARISON OF DEVICE UTILIZATION SUMMARY

Logic utilization	**Available**	**Used in [7]**	**Used in proposed work**
Number of slice flip flops	687,360	2663	2059
Number of slice LUT's	343680	1883	1662
Number of occupied slices	85920	722	689
Number of RAMB36E1	632	8	3
Number of DSP48E1'S	864	17	6

III. CONCLUSION

In this paper, we have proposed a FFT architecture which has less utilization of number of LUTs and Flip Flops compared with other architecture presented. By efficient addressing using only simple shift operations, we reduced number of clocks for reading and processing of address logic, achieved area and frequency optimized FFT core with minimum number of clock latency and maximum clock frequency (480 MHz).

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H.264 VIDEO CODEC FOR VIDEO TRANSMISSION OVER WIRELESS NETWORKS

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Abstract: Channel errors have a very detrimental effect on the perceptual video quality. Despite the research done in the field of wireless multimedia, delivery of real-time interactive video over noisy wireless channels is still a challenge for researchers. This project presents a method for improving the quality of video transport using H.264/AVC over wireless networks that is the prioritization of different parts of I and P-frames.. The video streaming and encoding model is implemented using Matlab Simulink model. The performance of the video streaming has been evaluated using JM 19.0(Joint Model) software. This paper focuses to improve the storage capacity and enhance the transmission rate by increasing the compression efficiency maintaining the same better video quality.

Key Words: H.264/AVC, JM, video streaming, transmission rate.

I. INTRODUCTION

Video has been an important media for communications and entertainment for many decades. Movie is a form of entertainment that enacts a story by screening a series of images giving the delusion of continuous movement. The trick was already known in second-century China, but remained inquisitiveness up to the end of the 19th century. The invention of motion picture camera around 1888 allowed the individual component images to be captured and stored on a single reel. For the first time, this has made possible the process of recording scenes in an automatic manner. Further to that, a hasty transformation occurred with the development of a motion picture projector to enlarge these moving picture shows onto a screen for an entire audience. [1]

II . METHODOLOGY

The part10 of MPEG-4 which is H.264 is compressed using the coding technique to eliminate the redundancy facilitating the fidelity of the video.[2] The compression method is carried out in the CODEC process.[3] The analysis has been done with the help of evaluating standard videos in MATLAB Computer Vision Toolbox. The conversion from AVI to YUV is done using MATLAB. The values of PSNR and MSE are calculated using JM software.

III. VIDEO STANDARD H.264 or MPEG-4 PART 10/AVC

MPEG-4 has about 30 parts, one for each technology. The two parts we are most concerned with are parts 10 and 14. Part 10, of MPEG-4 describes AVC or H.264. Many broadcast pipelines and distribution channels have adopted H.264 as well.H.264 is also expected to accelerate the adoption of megapixel cameras since the highly efficient compression technology can reduce the large file sizes and bitrates generated without compromising image quality. H.264 provides savings in network bandwidth and storage costs, it will require higher performance network cameras and monitoring stations.[4]

IV. DEVELOPMENT OF H.264

H.264 is the result of a joint project between the ITU-T‘s Video Coding Experts Group and the ISO/IEC Moving Picture Experts Group (MPEG). H.264 is the name used by ITU-T, while ISO/IEC has named it MPEG-4 Part 10/AVC since it is presented as a new part in its MPEG-4 suite. The MPEG-4 suite includes, for example, MPEG-4 Part 2, which is a standard that has been used by IP-based video encoders and network cameras.[5]

Designed to address several weaknesses in previous video compression standards, H.264 delivers on its goals of supporting: Implementations that deliver an average bit rate reduction of 50%, given a fixed video quality compared with any other video standard. Error robustness so that transmission errors over various networks are tolerated. Low latency capabilities and better quality for higher latency. Straightforward syntax specification that simplifies implementations.

Exact match decoding, which defines exactly how numerical calculations are to be made by an encoder and a decoder to avoid errors from accumulating.H.264 also has the flexibility to support a wide variety of applications with very different bit rate requirements.[6]

i) H.264 VIDEO CODING

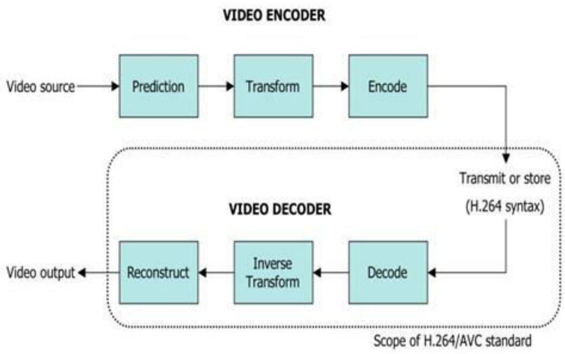
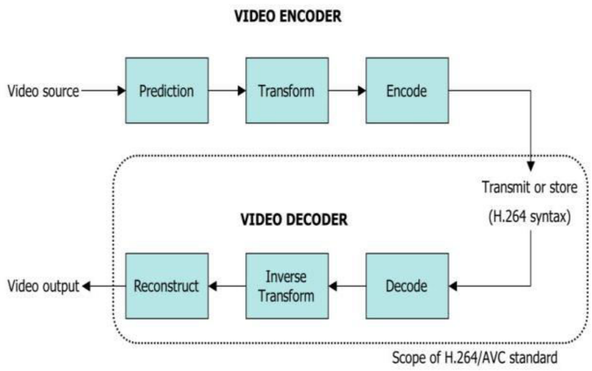
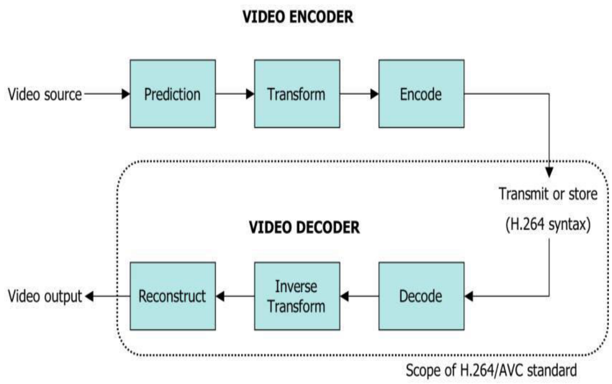


Figure 1: Block diagram of H.264





A video CODEC encodes a source image or video sequence into a compressed form and decodes this to produce a copy or approximation of the source sequence. If the decoded video sequence is identical to the original, then the coding process is lossless; if the decoded sequence differs from the original, the process is lossy. The CODEC represents the original video sequence by a model, an efficient coded representation that can be used to reconstruct an approximation of the video data. Ideally, the model should represent the sequence using as few bits as possible and with as high a fidelity as possible. These two goals, compression efficiency and high quality, are usually conflicting, i.e. a lower compressed bit rate typically produces reduced image quality at the decoder.[7]

ii) Overview of transform processes

The inverse transform and re-scaling or 'inverse quantization' processes shown in are defined in the H.264/AVC standard. These processes or their equivalents must be implemented in every H.264-compliant decoder. The corresponding forward transform and quantization processes are not standardized but equivalent processes can be derived from the inverse transform / rescaling processes.[8]

In an H.264 encoder, a block of residual coefficients is transformed and quantized. The basic transform, _core transform', is a 4×4or8×8integer transform, a scaled approximation to the Discrete Cosine Transform, DCT. In certain cases, part of the output of this integer transform is further transformed, 'DC transform', using a Hadamard Transform. The transform coefficients are scaled and quantized. The corresponding inverse processes. The DC inverse transform, if present, is carried out before rescaling. The rescaled coefficients are inverse transformed with a 4×4 or 8×8 inverse integer transform.[9]

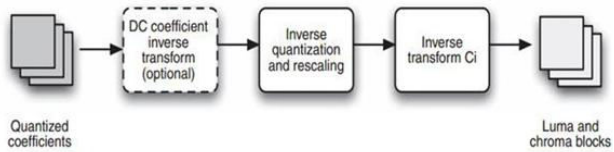


Figure2: Rescaling and inverse transform

iii) FRAMES:

Depending on H.264 profile, different types of frames such as I-frames, P-frames and B-frames may be used by an encoder. An I-frame or Intra frame is self-contained frame that can be independently decoded without any reference to other frames. The first image in a video sequence is always an I-frame. I-frames are needed as starting points of new viewers or synchronization points if the transmitted bit stream is damaged. I-frames can be used to implement fast forward, rewind and other random access functions. An encoder automatically inserts I-frames at regular intervals or on demand if new clients are expected to join in viewing a stream. The drawback of I-frames is that they consume much more bits, but on the other hand, they do not generate many artifacts. A P-frame, which stands for predictive inter frame, makes references to parts of earlier I and/or P-frames to code the frame. P-frames usually require fewer bits than I-frames, but a drawback is that they are very sensitive to transmission errors because of the complex dependency on earlier P and I reference frames.

A B-frame or bi-predictive inter frame, is a frame that makes references to both an earlier reference frame and a future frame.[10]

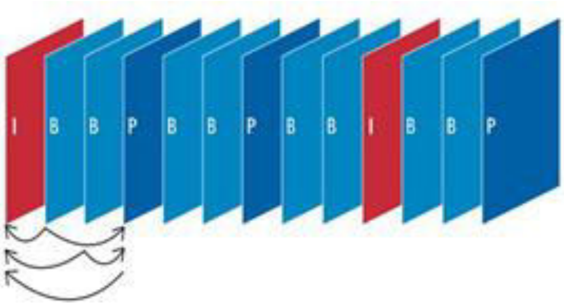


Figure3: Divisions of frames

RESULTS

Quality Metrics

Peak Signal to Noise Ratio (PSNR)

PSNR is the peak signal-to-noise ratio in decibels (dB). The PSNR is only meaningful for data encoded in terms of bits per sample, or bits per pixel. For example, an image with 8 bits per pixel contains integers from 0 to 255.

The following equation defines the PSNR:

$$20\log_{10}\frac{2^B-1}{\sqrt{MSE}} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots 1$$

Where, MSE represents the mean square error and B represents the bits per sample.

Mean Square Error (MSE)

The mean square error between a signal and image, X, and an approximation, Y, is the squared norm of the difference divided by the number of elements in the signal or image:

$$\frac{\|X-Y\|^2}{N} \quad \ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots\ldots \quad 2$$

Where, N is the number of elements in the signal or image.

Analysis of Video ENCODING AND DECODING Model

The analysis of the model is done using 9 basic steps. Step 1: Reading input video file

Step 2: Converting the video into the frames

Step 3: Converting the RGB frames to Grey images

Step 4: Applying the Discrete Cosine Transforms on Gray images

Step 5: Encoding the images

Step 6: Transmit and Store

Step 7: Decoding the images

Step 8: Applying the Inverse Discrete Cosine Transform Step

9: Reconstructing the Source video

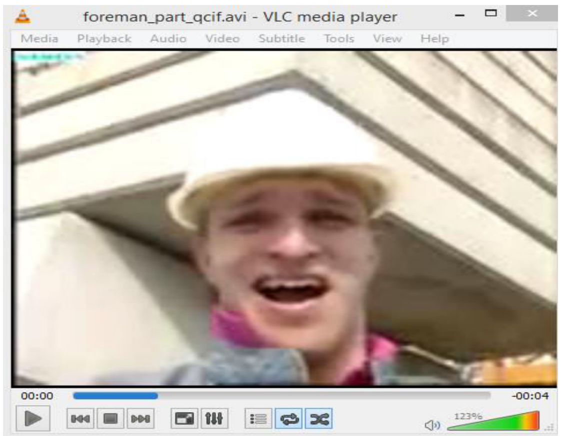


Figure4: Input video

Step2: Converting the input video into the residual frames

Figure5: converted video frames

The video is divided into frames with a frame rate of 30 frames/sec.

For step 3:

Converting the RGB frames to Grey images

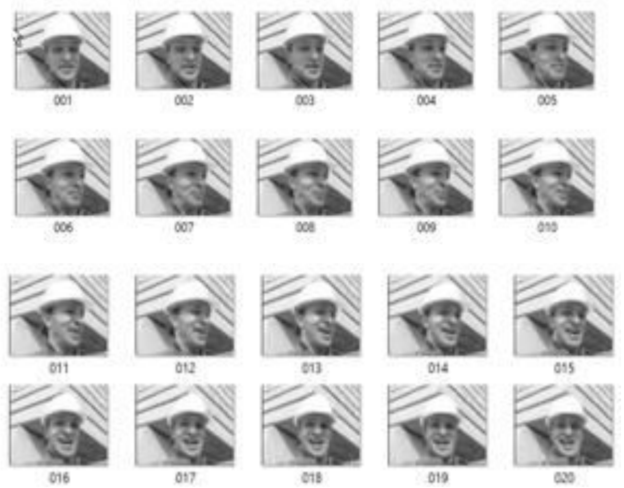


Figure6: converted gray frames

The H.264 works only on the monochromic image. Hence the video frames must be converted to gray images. The frames are converted with 8 bit depth to facilitate the transform and quantization.

For step4:

The grey scale images are applied as inputs to the transform. The transform implemented is the Discrete Cosine Transform. It is used as it requires less number of computations, low complexity. The image coefficients are calculated from the DCT and used in the quantization process to transmit the image frames into bits for effective transmission.

For step5:

Encoding the images

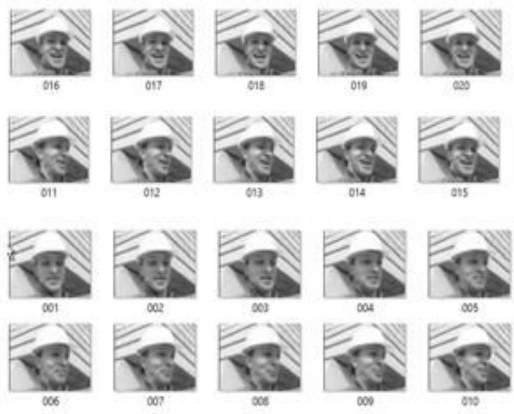


Figure7: encoding video frames

The transformed image frames are encoded so as to eliminate the redundancy in the data that is to be stored and transmitted. The bitrate can be evaluated after the video is encoded.

For step6:

Transmit and Store

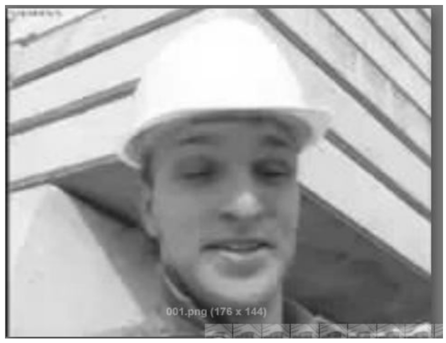


Figure 8: storing and transmitting video

The encoded video is stored and transmitted which is the compressed video format of H.264.

For step7:

Decoding the images

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Figure9: decoding the video

After the transmission the H.264 compressed video is decoded which is a reversible operation of the encoding process.

For step8:

Applying the Inverse Discrete Cosine Transform

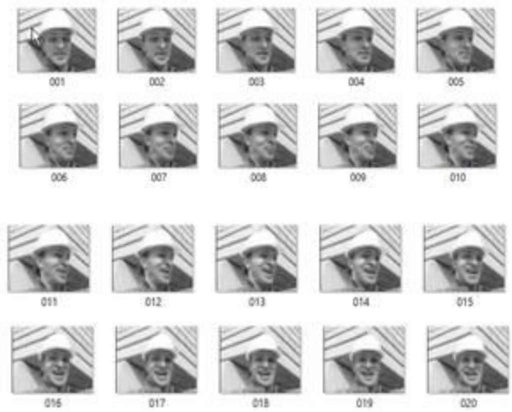

Figure10: reconstructed image frames

The decoded frames are reconstructed with the help of inverse Discrete Cosine Transform. The bit rate, PSNR values can be compared to the encoded frames to find the efficiency of the compressed standard.

For step9: Reconstructing the Source video

Figure11: output video

The reconstructed frames are finally clubbed into single video file which is the replica of the input source file.

Evaluated results:

Bit rate of the input video is 1380 Kbit/s

Encoding bit rate is 1328 Kbit/s

Decoding bit rate is 1360 Kbit/s

Table1: MSE and PSNR values of frames calculated using MATLAB

foreman_part_qcif.avi	**PSNR (dB)**	**MSE**
Input file	**38.9453**	**8.47**
Output file	**39.1569**	**8.14**

It is clearly evident from the above values that, despite lower bandwidth, there is a little variation in value of PNSR from input to output video. This explains the Robustness of the employed scheme for video streaming.

DETAIL DESCRIPTION OF PSNR VALUES OF INPUT VS OUTPUT

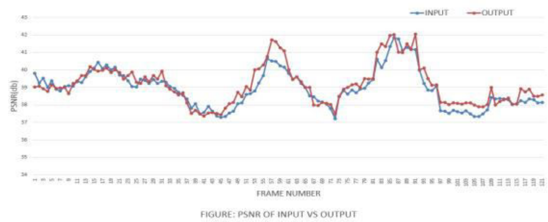

Figure11: PSNR input VS out put values

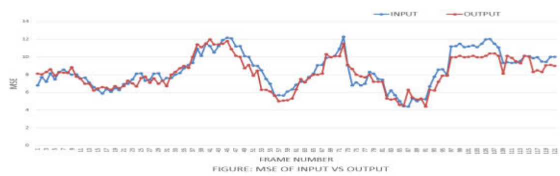

Figure12: MSE input VS output values

CONCLUSION

H.264 presents a huge step forward in video compression technology. It offers techniques that enable better compression efficiencies due to more accurate prediction capabilities, as well as improved resilience to errors. It provides new possibilities for creating better video encoders that enable higher quality video streams, higher frame rates and higher resolutions at maintained bit rates (compared C with previous standards), or, conversely, the same quality video at lower bit rates.

FUTURE SCOPE

In the need of the better video standards, compared to the previous standards H.264 ensures the better quality of video transmission with less error over the channels with high storage capacity. It implements a simple transform coding along with a flexible prediction method to isolate the frames. It also provides high latency and strength to the video that is to be transmitted. It facilitates the better ways of eliminating the redundancies in the codec format.

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DESIGN AND DEVELOPMENT OF SQUARE PATCH ANTENNA WITH 90^{O} HYBRID FEED FOR COMMUNICATION APPLICATIONS

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Abstract— **Microstrip antennas find wide application in high-speed vehicles, and missiles, tanks, satellite communications etc. The main advantage of these antennas over conventional microwave antenna is lightweight, low volume, low cost, planar structure and compatibility with integrated circuits. The present paper deals with the design and development of 90^{o} hybrid feed square patch stacked antenna for communication applications. The design of square patch and 90^{0} hybrid feed has been carried out at frequency of 3 GHz on epoxy glass substrate, the radiation pattern of the square patch has been experimentally studied. The effect of stacked patches placed above the square patch has been studied experimentally for different cases like 1,2,3 and 4 stacked patches placed one above other above the driven square patch. From the experimental result it has been found that performance of the case of 1 + 2 (one driven element and two parasitic element) is optimum with bandwidth of 16 % and VSWR 1.42 the performance degrades the no of practical elements is increased that is for case 1 + 3 and 1 + 4 etc., The performance of 1 + 2 case of also found to be superior to the performance 1+ 0 and 1+1 cases experimentally studied, also been carried out for cross Polarization and co – polarization.**

Keywords—Square patch antenna, hybrid feed, polarization etc.

INTRODUCTION

The microstrips find potential application in various diversified fields especially in high speed space vehicles, missiles, tanks and other strategic defense equipment's. Since the inception, microstrip antenna attracted attention of large number of researches world over which has given stimulus to the research and development resulting into many diverse applications such as aircrafts missiles, space vehicles, satellite communications, telemetry, radars and other defense equipments [1-6]. The microstrip structures radiating circularly polarized waves play an important role in communication because in such cases problem of alignment is completely removed. Variety of microstrip antenna is one of those, which can provide circular polarization with hybrid feed. In the present endeavor, therefore an attempt has been made to design and develop a square microstrip patch for circular polarization. Cavity model in one of the most widely used methods in analyzing the performance of microstrip antennas. In this model the region between the conducting patch and the ground plane is assumed to be a resonant cavity the main drawback in the cavity theory, i.e, perfect magnetic wall boundary condition, has been overcome in the fringing capacitance mode. In the fringing capacitance mode, the square patch backed by ground plane with a dielectric substrate in between is assumed to be approximating a capacitor. Whenever, the patch is excited the side of the patch is extended due to fringing field. The patch is considered as a capacitor which have to different values of capacitance for excited and unexcited conditions. Whenever the patch is excited the effect of fringing field increases the effective resonator area hence, the increases capacitance. The difference of the patch capacitance under excited and unexcited conditions provides the fringing field capacitance which forms the basis for the source responsible for the entire radiation [5-9]. The design square patch and 90^{0} hybrid feed has been carried out at frequency of 3.0 GHz, an epoxy glass substrate, the radiation pattern of the square patch has been experimentally studied. The effect of stacked patches placed the above the square patch has been studied experimentally for different cases. The structure of 90^{o} Hybrid feed is shown in Fig.1.

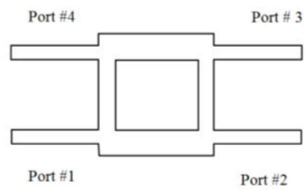

Fig.1 90^{o} Hybrid

SPECIFICATIONS

The specification of epoxy glass substrate:

Dielectric constant ($\in_r$) = 4.5

Height of the substrate (h) = 0.16cm

Thickness of the strip (t) = 0.001cm

Operating frequency (f) = 3.0 GHz

Free space velocity (C) = $3x10^{10}$ cm/sec.

Table 1: The width of the patch as shown in table given below

Strip impedance (Ω)	Effective dielectric constant ($\in_e$)	Widths (cm)
50	3.425	0.216
35.35	3.460	0.386

LENGTHS OF λ/4 SECTIONS

As the width is different for different impedance arms, so are their dielectric constants, hence the respective wavelengths are also different. The width of the patch and impedance of quarter wavelength which is shown in Table 1 and 2.

Table 2: The impedance of the quarter wavelength

Impedance of the section (Ω)	Quarter wavelengths (cms)
50	1.3508
35.35	1.3438

The completes the design of the microstrip patch antenna with a branch line 90^0 hybrid as shown in Fig.2.

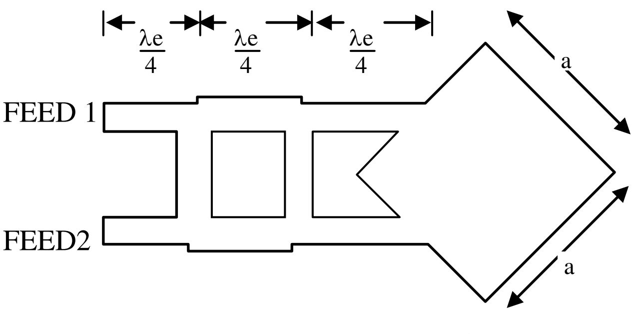

Fig. 2 Square Microstrip patch with 90° Branch Line Hybrid for CP Operation

DESIGN OF THE PATCH

The patch size has been calculated by using iteration method. The approximate patch dimension is determined as

$$a^1 = C/2f_0 \sqrt{\in_e} = 2.486$$

To account for the fringing field, the change effective dimension is given be 2Δ, where

$$a^1 = 0.00346$$

Thus $a = a^1 - 2\Delta = 2.5$cm

Hence the patch size is a x a = 2.5 x $2.5cm^2$

DEVELOPMENT OF ANTENNA

The first step is the tracing of the antenna shape on tracing paper, enlarged artwork is photo reduced using a high precession camera to produce a high resolution negative which later used for exposing the photo resist. The laminate is cleaned to ensure proper adhesion of the photo development process. The photo resist is now applied to both sides of laminate using laminator. After words, the laminate allowed to stand to normalize to soon prior to exposure and development.

The photographic negative is now held in very close contact using corner sheet of the applied photo resist to ensure the fine line resolution required with exposure to proper wavelength light. Polymerization of the exposed photo resist occurred, making it insoluble in the developer solution. The backside of antenna is completely exposed without mask since copper foil is returned to act as a ground plane, the protective corner sheet of the photo resist is removed and antenna is now developed in a developer which removes the soluble photo resist material. Then the antenna is etched visual inspection is used to assure proper etching [10-14].

EXPERIMENTAL INVESTIGATIONS

Experimental Measurements

The set – up was used for measuring the co-polar and cross-polar pattern of the antenna [2]. During the experiment, the output of the source was fairly constant. The source of the microwave power was quite stable and the frequency variations were negligible small. Isolator was used to avoid the reflection from the antenna. The receiver system was kept in the far zone ($2d^2 / \lambda$). Using the set up the radiation patterns of the antenna was measured. The data of measured radiation pattern using different number of parasitic elements are shown in Tables 3 and 4. Beam-width (degree) actual ration and gain was shown in Table 8 and 11. The data VSWR was shown in Table 6. Using the VSWR data, return-loss were we calculated. The data of return loss was shown in Tables 5 and 12. Calculated the bandwidth from the plot of VSWR, The data of band width was shown in Table 12. The data of axial ratio, maximum radiation is shown in Tables 7, 9, 10

Table 3: Data of Co-Polar radiation pattern for 90 hybrid –

feed square patch stacked antenna

Angle (Degree)	Relative Power (dB)				
	Driven element	Driven element +1 parasitic	Driven element +2 parasitic	Driven element +3 parasitic	driven element +4 parasitic
-90	-14.49	-12.78	-12.78	-13.7	-15.02
-80	-14.4	-10.36	-9.77	-11.28	-13.7
-70	-13.31	-10.56	-7.14	-12	-13.84
-60	-12	-8	-5.6	-9.38	-13.2
-50	-10.95	-6.1	-3.8	-8.33	-12.13
-40	-10.3	-4.92	-2.75	-6.95	-11.67
-30	-8.98	-3.93	-1.7	-6.03	-10.75
-20	-7.34	-3.28	-1.25	-5.25	-9.2
-10	-6.3	-2.49	-0.48	-4.46	-8.01
0	-5.6	-2	0	-4.07	-6.56
10	-6.49	-2.95	-0.483	-4.52	-7.15
20	-7.48	-3.75	-1.02	-5.64	-8.46
30	-9.31	-4.66	-1.9	-7.28	-10.23
40	-10.8	-6.03	-2.89	-8.4	-11.64
50	-12	-7.34	-4.46	-10.3	-12.66
60	-13.51	-9.25	-5.77	-11.2	-13.7
70	-14.03	-11.87	-8.98	-13.05	-13.7
80	-14.69	-11.93	-10.95	-12.46	-13.75
90	-14.62	-12.78	-12.13	-14.75	14.33

Table 4: Data for Cross - Polar radiation pattern 90° hybrid –feed square patch stacked antenna

Angle (Degree)	Relative Power (dB)				
	Driven element	Driven element +1	Driven element +2	Driven element +3	Driven element +4
-90	-17.4	-16.53	-16.45	-15.73	-17.69
-80	-16.36	-12.17	-11.26	-15.03	-17.76
-70	-16.71	-11.62	-11.05	-12.8	-16.15
-60	-14.06	-9.37	-7.69	-11.47	-15.1
-50	-12.37	-7.21	-5.45	-10.56	13 85
-40	-11.06	-6.05	-4.06	-8.88	-12.59
-30	-10.07	-5.2	-2.66	-8.04	-11.33
-20	-9.3	-4	-1.89	-6.85	-10.14
-10	-8.11	-3.13	-0.98	-5.52	-9.52
0	-7.06	-2.62	-0.45	-4.69	-7.9
10	-8.32	-3.36	-1.1	-5.73	-9.43
20	-9.86	-4.6	-2.13	-7.51	-10.84
30	-10.8	-6.36	-3.36	-9.23	-12.1
40	-13.08	-8.18	-4.8	-10.9	-14.2
50	-14.64	-11.38	-6.71	-13.32	-15.15
60	-15.58	-12.87	-10.4	-14.71	-16.02
70	-16.85	-11.96	-9.93	-15.38	-16.76
80	-16.64	-13.94	-12.87	-16.63	-17.48
90	-18.57	-16.38	-16.7	-17.22	-17.55

Table 5: Data of Return – loss for 90 ^{0}hybrid- feed square patch stacked Antenna

Frequency (GHz)	*Driven eleme*	*Driven eleme*	*Driven eleme*	*Driven element +3*	*Driven element +4*
2	-1.434	-1.51	-1.52	-1.38	-1.28
2.1	-1.7	-1.76	-1.86	-1.62	-1.55
2.2	-1.99	-2.12	-2.29	-1.85	-1.79
2.3	-2.49	-2.65	-2.87	-2.21	-2.12
2.4	-3.18	-3.49	-3.74	-2.92	-2.65
2.5	-3.76	-4.12	-4.37	-3.78	-3.38
2.6	-4.37	-5	-5.07	-4.59	-4.18
2.7	-5.33	-6.46	-7.35	-5.61	-5.04
2.8	-6.72	-8.84	-10.35	-7.85	-6.36
2.9	-8.8	-11.28	-12.73	-10.16	-7.7
3	-11.5	-13.84	-15.2	-12.62	-10.58
3.1	-8.8	-10.88	-12.51	-9.84	-7.63
3.2	-6.15	-8	-9.31	-7	-5.26
3.3	-4.87	-5.95	-6.81	-5.43	-4.21
3.4	-3.67	-4.43	-5	-4.1	-3.25
3.5	-2.92	-3.39	-3.67	-3.13	-2.68
3.6	-2.49	-2.78	-2.92	-2.65	-2.29
3.7	-2.11	-2.43	-2.53	-2.27	-1.93
3.8	-1.82	-1.99	-2.12	-1.89	-1.72
3.9	-1.66	-1.76	-1.82	-1.7	-1.62
4.0	-1.41	-1.53	-1.59	-1.5	-1.34

TABLE 6: DATA OF VSWR FOR 90° HYBRID - FEED SQUARE PATCH STACKED ANTENNA

Frequency (GHz)	*Driven element*	*Driven element +1 parasitic*	*Driven element +2 parasitic*	*Driven element +3 parasitic*	*Driven element +4 parasitic*
2	12.14	11.48	11.4	12.6	13.5
2.1	10.2	9.88	9.35	10.7	11.2
2.2	8.76	8.2	7.6	9.4	9.69
2.3	7	6.6	6.1	7.9	8.2
2.4	5.51	5.04	4.71	6	6.6
2.5	4.69	4.29	4.05	4.66	5.2
2.6	4.05	3.52	3.23	3.87	4.23
2.7	3.36	2.81	2.5	3.2	3.54
2.8	2.71	2.13	1.91	2.36	2.85
2.9	2.12	1.75	1.6	1.9	2.4
3	1.72	1.51	1.42	1.61	1.84
3.1	2.12	1.8	1.62	1.95	2.42
3.2	2.94	2.3	2.04	2.6	3.4
3.3	3.66	3.03	2.68	3.3	4.2
3.4	4.8	4	3.54	4.3	5.4
3.5	6	5.18	4.8	5.6	6.53
3.6	7	6.3	6	6.6	7.6
3.7	8.25	7.17	6.9	7.68	9
3.8	9.55	8.76	8.2	9.2	10.12
3.9	10.48	9.88	9.55	10.2	10.7
4.0	12.28	11.34	10.94	11.6	12.92

TABLE 7: DATA OF AXIAL RATIO FOR 90^0 HYBRID – FEED SQUARE PATCH STACKED ANTENNA

Angle (Degree)	*Driven element*	*Driven element +1 parasitic*	*Driven element +2 parasitic*	*Driven element +3 parasitic*	*Driven element +4 parasitic*
-90	-2.91	-3.75	-3.67	-2.03	-2.67
-80	-1.96	-1.81	-1.49	-3.75	-4.06
-70	-3.4	-1.06	-3.9	-0.8	-2.31
-60	-2.06	-1.37	-2.09	-2.09	-1.9
-50	-1.42	-1.11	-1.65	-2.23	-1.72
-40	-0.76	-1.13	-1.31	-1.93	-0.92
-60	-1.09	-1.27	-0.96	-2.01	-0.58
-20	-1.96	-0.72	-0.64	-1.6	-0.94
-10	-1.81	-0.64	-0.5	-1.06	-1.51
0	-1.46	-0.62	-0.45	-0.62	-1.34
10	-1.83	-0.41	-0.617	-1.21	-2.28
20	-2.38	-0.85	-1.11	-1.87	-2.38
30	-1.49	-1.7	-1.46	-1.95	-1.87
40	-2.28	-2.15	-1.91	-2.5	-2.56
50	-2.64	-4.06	-2.25	-3.02	-2.49
60	-2.07	-3.62	-4.63	-3.51	-2.32
70	-2.82	-0.09	-0.95	-2.33	-3.06
80	-1.95	-2.01	-1.92	-4.17	-3.73
90	-3.95	-3.6	-3.93	-2.47	-3.22

TABLE 8: DATA FOR BEAM-WIDTH (DEGREE)

No. of Elements in 90^0 Hybrid - feed square patch stacked antenna	*Beam-width (Degree)*	
	Co-Polar Plane	Cross-Polar Plane
1+0	59	56
1+1	79	64
1+2	98	65
1+3	69	60
1+4	50	53

TABLE 9 : DATA FOR MAX. RADIATED POWER(dB)

No. of Elements in 90° Hybrid - feed square patch stacked	*Maximum radiated power(dB)*	
	Co-Polar Plane	Cross-Polar Plane
1+0	-5.6	-7.06
1+1	-2	-2.62
1+2	0	-0.45
1+3	-4.07	-4.69
1+4	-6.56	-7.9

TABLE 10: DATA FOR AXIAL RATIO(dB)

No. of Elements in 90 Hybrid - feed square patch stacked antenna	*Axial ratio (dB)*
1+0	-1.46
1 + 1	-0.62
1+2	-0.45
1+3	-0.62
1+4	-1.34

Table 11 : Data for Gain (dB)

No. of Elements in 90 Hybrid - feed square patch stacked antenna	*Gain (dB)*
1+0	8.95
1 + 1	7.11
1+2	6.10
1+3	7.97
1+4	9.91

TABLE 12: DATA FOR RETURN – LOSS (dB)

No. of Elements in 90 Hybrid - feed square patch stacked antenna	*Return-loss (dB)*
1+0	-11.5
1 + 1	-13.84
1+2	-15.2
1+3	-12.62
1+4	-10.58

TABLE 13: DATA FOR VSWR

No. of Elements in 90 Hybrid - feed square patch stacked antenna	VSWR
1+0	1.72
1 + 1	1.51
1+2	1.42
1+3	1.61
1+4	1.84

Table 14: Data for Band – Width

No. of Elements in 90 Hybrid - feed square patch stacked antenna	Band – Width (%)
1+0	5.33
1 + 1	10.66
1+2	16
1+3	8
1+4	2.66

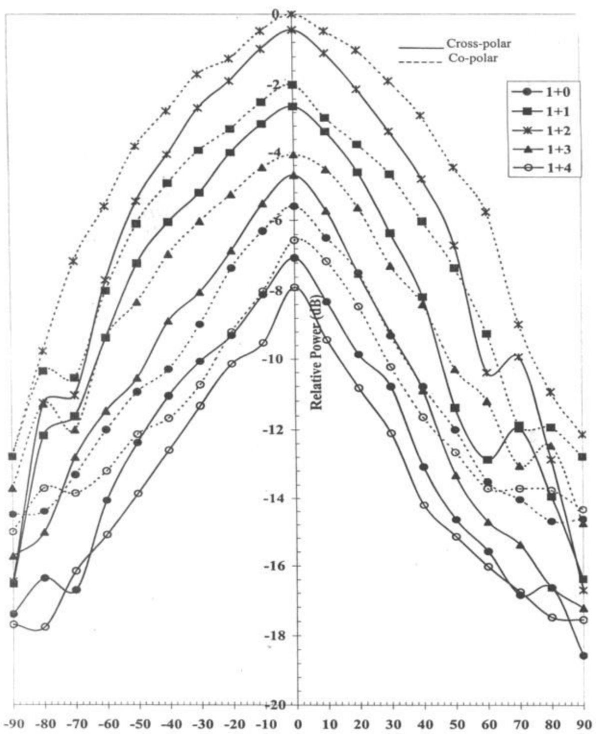

Fig.4. Radiation pattern of 90^{o} hybrid-feed square patch stacked antenna for co-polar and cross-polar plane

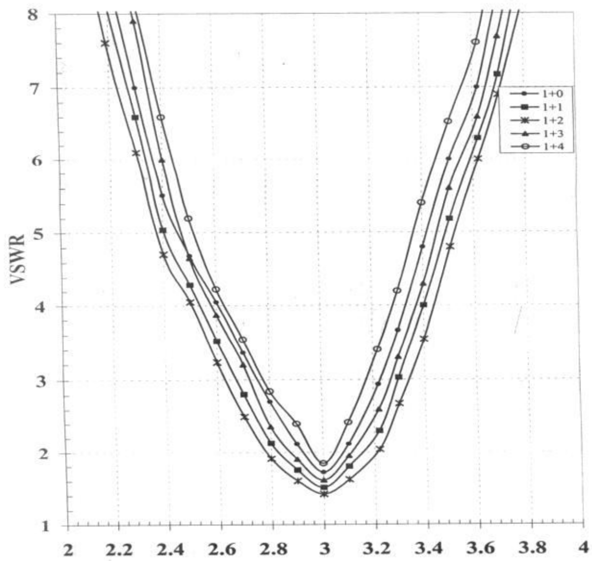

Fig.5. Variation of VSWR with Frequency for 90^{o} hybrid-feed square patch stacked antenna

RESULTS AND DISCUSSION

Measurements were made for various parameters such as co – polar and cross –polar radiation pattern, VSWR, Return – loss and Max. Radiated power and Axial – ratio which is shown in Tables from 3 to 14. It is found that the Radiated power is maximum for the antenna with 2 parasitic elements. Initially the radiated power increase with parasitic elements and becomes maximum for 2 parasitic elements and decreases if number of parasitic element is increased. This is also observed from the VSWR data and return loss. VSWR is obtained at 3 GHz for the cases of optimum 2 parasitic element and return loss of found to be less than -15 dB. The variation of the Radiated power for co–polar and cross – polar cases is shown in Fig.4 and VSWR is shown in Fig. 5. It is observed that radiation increases with parasitic element and becomes maximum for 2 parasitic elements loaded patch. The radiated power decreases with increasing value of parasitic elements. The Axial ratio is found to be maximum for 3 parasitic elements antenna and value falls with increasing and decreasing no. of parasitic elements. However the axial ratio with the limits and radiation remains in circular – polarization.

CONCLUSION

A 90 ^{0} hybrid – feed square patch stacked antenna has been designed on epoxy glass substrate. The patches are square patch, for feeding, a suitable point on patch is marked where input impedance is 50 ohm. The antenna is developed by using photolithographic process.

From the experimental results it has been found that the performance of the square patch stacked with 2 parasitic elements is found to be optimum.

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VIDEO SHOT BOUNDARY DETECTION USING SUPPORT VECTOR MACHINE

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Abstract—**Video shot detection is an important contemporary problem since it is the first step toward automatic indexing, content based video retrieval and many other different applications. A novel shot boundary detection using wavelet and Support Vector Machine is proposed in this paper. Shot boundary detection algorithms work by extracting the color and the edge in different direction from wavelet transition coefficients. A multi-class support vector machine (SVM) classifier is used to classify the video shot into three categories: cut transition (CT), gradual transition (GT) and normal sequences (NF). To enhance the robustness of the algorithm, we form the feature vector from all frames within a temporal window. Numerical experiments using a variety of videos demonstrate that our method is capable of accurately detecting and discriminating shot transitions in videos with different characteristics.**

Keywords-Shot Boundary Detection; Support Vector Machine; DWT;

II. INTRODUCTION

Recent advances in the decreasing storage costs, increasing processing power, and the growing availability of broadband data connection, have led to the widespread use and availability of digital videos. Digital video is a rich and subjective source of information. So searching for the video sections in which we are interested is a very difficult task. Therefore, the demand of new technologies and tools for effective and efficient indexing, browsing and retrieval of video data has been exacerbated by recent trends. The first step in retrieval video databases is shot boundary detection. A shot is defined as a sequence of frames taken by a single camera with no major changes in the visual content. According to whether the transition between shots is abrupt or gradual, the shot boundaries can be categorized into two types: cut transition (CT) and gradual transition (GT) [1]. To segment a video shot, many efforts have been devoted into this area for the past years, and many different methods have been proposed [2][3]. Almost all shot change detection algorithms reduce the large dimensionality of the video domain by extracting a small number of features from one or more regions of interest in each video frame [4]. In this paper, we propose an algorithm for shot transition detection with different features. We choose the color and the edges in the vertical, the horizontal and the diagonal direction as the feature vector in my shot detection system.

To reduce the detection time, the algorithm uses the wavelet to extract the features. Traditionally, video shot segmentation approaches rely on thresholding method, which do not generalize well since characteristics is different for the different videos. In this paper, a multi-class support vector machine (SVM) classifier is constructed to classify the video frames within a sliding window into cut transition, gradual transition and normal sequences. The testing result of the experiment shows that the method has good accuracy for shot boundary detection.

II. FEATURE SELECTION

A. *Wavelet*

Wavelet is a nice tool to decompose an image signal into sub bands. Not only can it give the desired low frequency and high frequency information we need to extract our feature parameters, but also it is fast and easy to compute and requires only linear time in the size of the image and very little code [5]. DWT decompose an image into four sub bands, which is the low frequency (LL), the vertical high frequency

(LH), the horizontal high frequency (HL) and the diagonal high frequency (HH) respectively. The low frequency looks very similar to the original image but the size is a half of the original one through a down sampling.

The vertical high frequency includes the information about a vertical element of the original image, which depicts the vertical directional edge of the original image. Similarly the horizontal high frequency is the horizontal directional edge of the original image. The diagonal high frequency corresponds to the diagonal directional edge of the original image. In this case, large color and edge features can be captured from the DWT image.

B. Feature mining based on wavelet

To define whether two frames are separated with an abrupt cut or a gradual transition we have to look for a difference measure between frames. It is hard to use one single feature to capture all the characteristics of all kinds of shot transitions efficiently because different shot transitions have different characteristics. In our approach we use multi features. The color and the edges in the vertical, the horizontal and the diagonal direction are chose as the feature vector in my shot detection system. The color and edges difference between the two consecutive frames is very large in the cut transition. There are a series of local larger difference where gradual change is occurring. The local motion, camera motion and the illumination changes often cause some misses on shot change detection. The color is sufficiently variant to illumination changes and motion, but the edge is not. The color difference between the two consecutive frames has a maximum value but the edge difference value is lesser where the illumination changes or slow motion is happening. Thus we can eliminate false detections that are caused by illumination changes and slow motion.

We partitioned a frame into 32×32 equal size blocks, and each block is decomposed through DWT in one levels. DWT decompose each block into four subbands. The variation of the color features between two frames can be calculated according to LL bands of each decomposed block. The difference of the edge in the vertical, the horizontal and the diagonal direction of two frames can be computed depending on the HL, LH, and HH bands of each decomposed block respectively. The algorithm is depicted as follows:

1). Color Difference

Let E(l) denote the total of the low frequency coefficient of the block L, which can be computed by following formula.

$$E(l) = \sum_{k=1}^{M} C_l(k) \qquad (1)$$

Where $C_l(k)$ is the coefficient value, M represents the total number of low the frequency coefficient of the block

l. Color difference of block l between two frames can be calculated by the following formula.

$$d_t(l) = E_t(l) - E_{t+1}(l) \qquad (2)$$

In order to eliminate smooth intervals, produce a new sequence named $d_t'(l)$ according to the following formula.

$$d_t'(l) \begin{cases} o & d_t(l) < T_d \\ 1 & otherwise \end{cases} \qquad (3)$$

Finally the difference between two frames based on their color feature is given from the following equation:

$$D_t^{LL} = \sum_{l=1}^{N} d_t'(l) \qquad (4)$$

In which, *N* represents the total number of block of a frame.

2). Edge Difference

The same as the above algorithm, the difference between two frames based on their edge feature in the vertical, the horizontal and the diagonal direction can be obtained according to formula (1) − (4) , which denote by D_t^{HL} D_t^{LH} D_t^{HH} respectively.

C. Sliding Window

It is intuitively clear that comparison with neighborhoods in temporal dimension plays an important role in deciding the class of a particular frame. To fully account for this, a sliding window is applied to a number of past and future frames of a frame. The feature vector of the current frame is then constructed with the feature data of the frames in the sliding window. Figure2 displays clearly the Cut transition in the sliding window. The variance curve of the color and the edge difference in the Cut region forms a clear peak. Figure3 shows the gradual transition in the sliding window. There are a series of local larger value where gradual change is occurring. In Figure2 and Figure3, the smooth beeline intervals depict constant sequence. The color difference between the two consecutive frames has a maximum value but the edge difference value is lesser where the illumination changes or slow motion is happening (seen in Fig.4). Thus we can eliminate false detections that are caused by illumination changes and slow motion.

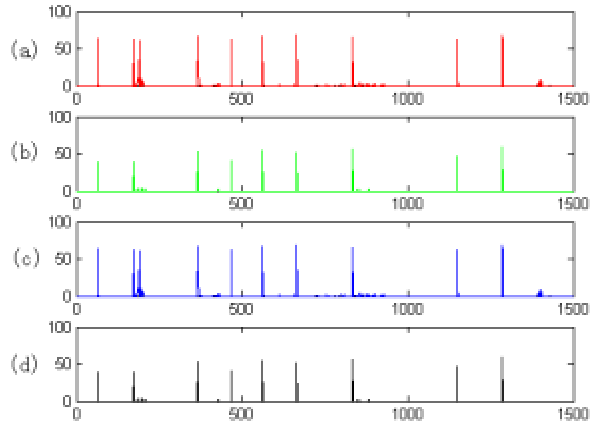

Figure 1. a sequence consists only cuts

III. SUPPORT VECTOR MACHINE CLASSIFIER

Having obtained the features of the continuity signal, the shot boundaries are detected by thresholding scheme as what most of the existing methods do. However, the thresholding method has several difficulties in achieving satisfactory results. First, the chosen threshold usually highly depends on the genres of videos. Second, a single threshold can not make full use of the contextual information of videos [1]. Therefore, we selected a classifier to identify the shot boundary detection. As for the selection of classifiers, SVM is preferred, not only for its solid theoretical foundations but also for its various empirical successes. In the following, we will introduce how to apply SVM to implement the shot boundary detection.

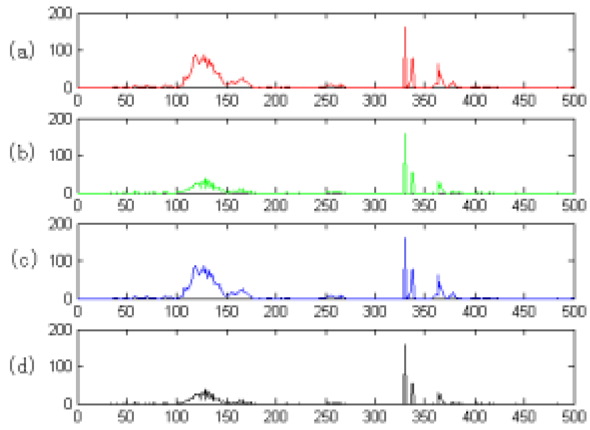

Figure 2. a sequence with gradual transitions

D. SVM

SVM (Support Vector Machine) is a useful technique for data classification, which based on the concept of the structural risk minimization using the Vapnik Chervonenkis (VC) dimension. A classification task usually involves with training and testing data which consist of some data instances. Each instance in the training set contains one”target value” (class labels) and several”attributes” (features). The goal of SVM is to produce a model which predicts target value of data instances in the testing set which are given only the attributes [7].

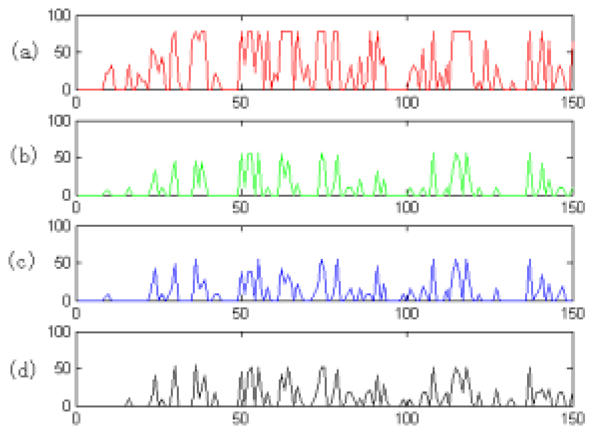

Figure 3. a sequence with illumination changes

E. SVM Classifier

We categorize shot boundary in three categories: normal sequences, abrupt cuts and gradual transitions. Nevertheless, the SVM is originally designed for binary classification. In our application, we have a three-class problem, thus we used the”one-against-one” approach in which for a k-class

Problem, k (k−1)/2 binary classifiers are constructed and each one is trained to discriminate data from two classes. In the following, we will describe how to apply"one-against-one" SVM to implement the shot boundary detection. (1). Let a set of training sample which can be obtained by human observer identifying.

$$F = \{(x_1, y_1), \cdots, (x_l, y_l)\} \in (X, Y)^l \qquad (5)$$

where x_i is input feature vector. The feature vector of

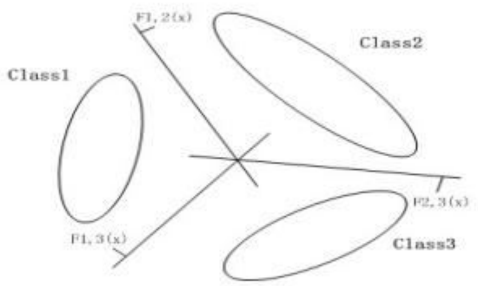

Figure 4. one against one method

the current frame is constructed with the feature data of the frames in the window. Suppose that the size of the window is n. The total number of features for a particular frame is then 4(n − 1) since there are four features per frame. Here we select n as 15. There are fifty six-dimensional feature vector as the input vector of SVM.

$$x_i = \quad (D_{t-7}^{LL}, \cdots, D_t^{LL}, D_t^{LH}, D_t^{HH}, D_t^{HL}, \cdots, D_{t+7}^{HH}) \in R^{56} \qquad (6)$$

In formula (8), y_i is output vector, y_i ∈(NF, CT,GT). If we assume that class label1corresponds to normal sequences, class label 2 to hard cuts and class label 3 to gradual transitions, three binary classifiers discriminating between pairs of classes(1,2), (1,3)and (2,3)are constructed. Figure4 shows the theory of a 3-class SVM classification. The decision function of classifier (i, j) is:

$$Min \quad \frac{1}{2}(w^{ij})^T w^{ij} + C\sum_{jt}^{l} \xi_t^{ij}$$

$$s.t. \quad \begin{cases} (w^{ij})^T \phi(x_t) + b^{ij} \geqslant 1 - \xi_t^{ij}, & if\ y_t = i \\ (w^{ij})^T \phi(x_t) + b^{ij} \geqslant -1 + \xi_t^{ij}, & if\ y_t = j \\ \xi_t^{ij} \geq 0 \end{cases}$$

In the final classification we use a voting strategy where the decision of each binary classifier is considered as a vote for its proposed class and the class with the maximum number of votes is selected. We join the result of SVM classification and get a series of frame labels which indicate NF, CF and GF, separately. Then the cut transition, gradual transition and normal sequences can be detected.

IV. EXPERIMENTALRESULTS AND DISCUSSIONS

The performance of the proposed algorithm is evaluated on the TREC-2001 video data set (V1) [8] and the four action movies (V2). There are a total of 198000frames, which includes cuts and dissolve, wipe, fade out/in (FOI)

Table I
VIDEOS FOR THE EXPERIMENT

Video	Frame	CT	GT
V1	108000	630	27
V2	90000	380	160

Table II
CT COMPARISON WITH THE ZHANG ALGORITHM

Video	recall	precision	F_1
our method	92.4 %	95.4 %	93.9 %
ZHJ method	79.7 %	83.0%	81.3%

gradual transition. Table 1 lists all the videos for the experiment. For each sequence, a human observer identifies the shot boundaries as the ground truth. The performance of a shot transition detection algorithm is usually measured with terms of recall and precision. The recall and precision are defined as following:

$$recall = \frac{N_c}{N_c + N_m} \times 100\% \qquad (7)$$

$$precision = \frac{N_c}{N_c + N_f} \times 100\% \qquad (8)$$

$$F_1 = \frac{2 \times precision \times recall}{precision + recall} \times 100\% \qquad (9)$$

where, N_c is the number of correct detections, N_m is the number of missed detections, N_f is the number of false detections. A good shot transition detector should have both high precision and high recall. F1 is a commonly used metric that combines precision and recall. Firstly, we should decide some parameter in the experiment. For SVR, we use the software Libsvm provided by the National Science Council of Taiwan to do SVM classification [9]. We have tested the performance on different kernels and find that the"RBF" kernel outperforms others. There are two parameters while using RBF kernels: the penalty parameter C and γ. It is not known beforehand which C and γ are the best for one problem. We adopt the cross-validation method to obtain the C and γ in this paper.

In the experiments, we compare our method with the other methods. Table2 and table3lists the performance of the proposed algorithm compared with Zhang algorithm [6], which used the twin comparison for shot transition detection. From the experimental results, we can see that the performance of our method is better than the Zhang algorithm. Zhang algorithm, only used the comparison of the pixels difference as the feature and it is not sufficient for shot

Table III

GT COMPARISON WITH THE ZHANG ALGORITHM

Video	recall	precision	F_1
our method	82.1 %	87.2 %	85.1%
ZHJ method	75.0 %	75.8 %	75.4 %

transition detection. Our algorithm uses the multi features and multi-class SVM to identify shot boundary detection and achieves better overall performance.

V. CONCLUSION

We have presented our complete framework of a video shot transition detection methodology. Unlike previous approaches which mainly rely on single feature to try to detect shot transitions, our method detects and identifies shot transitions with different features. By"one-against-one" SVM classifier, the videos shot are classified into hard cuts, gradual transitions and normal sequences. The testing result of the experiment shows that the method has good accuracy for shot boundary detection.

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CLUSTERING AND CLASSIFICATION OF IMAGES USING ABC-KFCM AND NEURAL NETWORK CLASSIFIER

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Abstract—**This paper presents a hybrid clustering algorithm and feed-forward neural network classifier for land-cover mapping of trees, shade, building, grass and road. It starts with the single step preprocessing procedure to make the image suitable for segmentation. The pre-processed image is segmented using the hybrid genetic-Artificial Bee Colony (ABC) algorithm that is developed by hybridizing the ABC and KFCM to obtain the effective segmentation in satellite image and classified using neural network. The performance of the proposed hybrid algorithm is compared with the algorithms like, Artificial Bee Colony(ABC) algorithm and KFCM .**

Keywords—ABC-KFCM; Segmentation Algorithm;Neural Network; Feature Extraction; Satellite Image Classification

INTRODUCTION

Image segmentation is a critical step of image analysis. The task of image segmentation can be stated as the clustering of a digital image into multiple meaningful non-overlapping regions with homogenous characteristics according to some discontinuity or similarity features like intensity, color or texture [1,2].

Depending on the way to deal with uncertainty about the available data, the clustering process can be categorized as Hard clustering or Fuzzy clustering. A hard clustering algorithm partitions the dataset into clusters such that one object belongs to only one cluster. This process is inappropriate for real world dataset in which there are no clear boundaries between the clusters. Since the inception of the fuzzy set theory thanks to Zadeh' work [3], researchers incorporate the concept of fuzzy within clustering techniques o handle the data uncertainty problem. The goal of unsupervised fuzzy clustering is to assign each data point to all different clusters with some degrees of membership.

The iterative unsupervised Fuzzy C-Means (FCM) algorithm is the most widely used clustering algorithm for image segmentation [4]. Its success is mainly attributed to the introduction of fuzziness about the pixels'membership to clusters in a way that postpones decision making about hard pixels' membership to latter.

Satellite image categorization field is quiet a challenging job. Recently, researchers have used different types of classification methods for enhancement of efficiency. Satellite image classification, an upgraded biologically motivated theory was applied by Lavika Goel [5]. This paper related to a study of their hybrid intelligent classifier along with other current Soft Computing classifier like: 1) Ant Colony Approach, 2) Hybrid Particle Swarm Optimization-cAntMiner (PSO-ACO2), 3) Fuzzy sets, 4) Rough-Fuzzy Tie up; the Semantic Web Based Classifiers and the traditional probabilistic classifiers such as the Minimum Distance to Mean Classifier (MDMC) and the Maximum Likelihood Classifier (MLC).

A hybrid Biogeography Based Optimization (BBO) based algorithm, which is an excellent land cover classifier for satellite image has been introduced by Navdeep Kaur Johal *et al.* [6]. Parminder Singh *et al.* have presented a FPAB/BFO based algorithm for the categorization of satellite image [7]. They intend to utilize the technique of Bacterial Foraging Optimization in order to categorize the satellite image.

M. Ganesh and V. Palanisamy have approached a method known as multiple-kernel fuzzy clustering (MKFCM) for satellite image segmentation [9].

This paper proposes a hybrid clustering algorithm and neural network classifier for satellite image classification.

In this work:

ABC algorithm and FCM are combined to improve the segmentation of images.

This paper is structured as follows: Second section delineates proposed technique, third section discusses analysis and the fourth section is conclusion.

PROPOSED ABC-KFCM AND NEURAL NETWORK

This section explains the proposed satellite image classification based on hybrid ABC-KFCM algorithm and Feed-Forward Neural Network. The classification would be done based on building, road, shade and tree. The proposed technique is discussed in two phases which are training and testing phases.

A.Training Phase

In training phase different colors of building , road, shade ,grass and trees are taken. For each different layer are extracted. For each layer histogram, maximum value of histogram and mean values are calculated. These features are given to neural network classifier to train the different regions in the image.

B. Testing Phase:

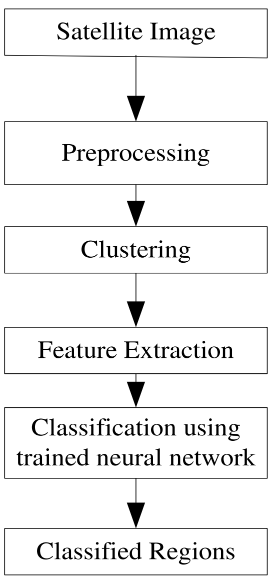

Fig.2 Process of Testing Phase

The Fig.2 explains as follows: the input satellite image is given for pre-processing using median filtering technique and the output of median filter is used for segmentation. In segmentation process, initially the H, T and L layers are extracted from the pre-processed image and the layers are given separately to the ABC-KFCM algorithm to cluster it. Thereafter, the clustered layers are merged one another and the feature extraction process is used on each merged clusters and then the extracted feature values of each merged clusters are applied to the trained neural networks to classify the building, road, shade and tree regions of the given input satellite image.

C.Clusteirng

In this process, the input pre-processed satellite image is converted different layers. i.e H, T and L layers are extracted from it. Thereafter, the ABC-KFCM algorithm is applied on each layer (H, T and L) separately to cluster the pixels. Here, the KFCM operator is incorporated in the ABC algorithm to segment the satellite image effectively. The segmentation process is done based on ABC-KFCM algorithm. Consider the ABC-KFCM algorithm is applied on H layer. The process is explained as follows: initially fixed numbers of initial solutions (food sources) are generated randomly by giving lower bound and upper bound. Each solution would contain centroids based on the required number of clusters.

After initial solutions are generated, the fitness is calculated for each solution. The calculation of fitness is as follows: initially the centroids in each solution are taken for clustering process and the clustering is done based on the minimum distance. The fitness is then calculated based on the equation given below:

$$fit_i = \sum_{j=1}^{J}\sum_{a=1}^{A}\left\|\left(x_a - C_j\right)\right\|^2$$

In the above equation fit_i denotes the fitness of i^{th} solution, where i=1,2 ---J; and x_a denotes a^{th} pixel x in j^{th} cluster; and j=1,2 ---J ; A is the total number of pixels in j^{th} cluster, where; and Cj denotes the centroid C of j^{th} cluster.

Employed Bee Operation

The employed bee then makes modification on the solution in its memory based on the local visual information and then calculates the nectar amount (fitness) of the new solution. If the nectar amount of the new solution is better than the old one, the bee would memorize the new one and forgets the old one. Otherwise it would keep the position of the old one in its memory. The employed bee operation is performed on each solution. To produce a candidate food position from old one in memory, the ABC uses the following expression:

$$S_{ij}^{new} = S_{ij} + \phi_{ij}\left(S_{ij} - S_{kj}\right)$$

In the above equation, $k \in \{1,2,....,I\}$and $j \in \{1,2,....,J\}$are randomly chosen index. Though k is determined randomly, it is different from i. i.e. Sij denotes the j^{th} centroid of i^{th} solution S; and S_{kj}

denotes the j^{th} centroid of k^{th} solution S. The Φij in the above equation is a random number between (-1,1) and it controls the production of neighbor food sources around S_{ij} and represents the comparison of two food positions visible to a bee. Using the above equation the j^{th} centroid of i^{th} solution S would get altered. We can alter two or more centroids based on the above equation and we would get a new solution. From the above formula the perturbation on the position Sij decreases as the difference between the parameters Sij and Skj decreases. Therefore the step length is adaptively reduced as the search approaches to the optimum solution in the search space. After the employed bee operation is performed on each solution, the fitness is calculated for each newly formed solution. If the nectar amount of the newly formed solution is better than the old one, the employed bee would memorize the new one and forgets the old one. The employed bees then share the nectar (fitness) information with the onlooker bees on the dance area.

Onlooker Bee Operation

The onlooker bee then evaluates the nectar information taken from all employed bees and chooses a food source with a probability to its nectar amount. The probability value is calculated for each solution and it calculated by the following equation:

$$\Pr_i = \left(\frac{0.25}{\max(fit)}\right) \times fit_i + 0.1$$

In the above equation $\Pr_i$ is the probability of i^{th} solution; max(fit) is the maximum fitness value among all the solutions; and fit_i is the fitness value of i^{th} solution. After calculating the probability of i^{th} solution, the onlooker bee would check whether $\Pr_i > rand$, where *rand* is a randomly generated number between zero and one. If it so, the onlooker bee would produce a new solution instead of this i^{th} solution. The new solution is formed based on the operation performed by the employed bee i.e. based on S_{ij}^{new} calculation. Then it would calculate the fitness (nectar amount) for the newly generated solution and compare with the old one. If the fitness of the newly formed solution is better than the old one, it would memorize the new one and forgets the old one.

Scout Bee Operation

The food source of which the nectar is abandoned by the bees is replaced with a new food source by the scouts i.e. the solutions which are not altered by any one of operations (which are employed bee operation and onlooker bee operation) is replaced by a new solution using scout bees. Consider i^{th} solution is not altered using either of employed bee operation and onlooker bee operation, the scout bee operation is performed on the i^{th} solution as defined below:

$$S_i^j = S_{\min}^j + rand(0,1)\left(S_{\max}^j - S_{\min}^j\right)$$

In the above equation S_{ji} is the j^{th} centroid of i^{th} solution; $S_{\min}^j$ is the minimum j^{th} centroid value among all the solutions; rand(0,1) is the random value between 0 and 1; and $S_{\max}^j$ is the maximum j^{th} centroid value among all the solutions. The scout bee operation is performed only if there has any abandoned solution. ABC operation is repeated until the iteration number set and a solution that has best fitness in the final iteration is taken for the KFCM operation.

C. Classification

The clustered regions features are calculated and compared with neural network classifer features. The best clustering algorithm is decided by the following parameters.

RESULTS AND DISCUSSION

This section delineates the results obtained for our proposed technique compared with the existing segmentation techniques. The performances are compared in terms of external metrics and internal metrics. The external metric is sensitivity performs the evaluations based on ground truth. The internal metric is Davies–Bouldin (DB) index.

A.Evaluation Metrics

The metrics used for evaluation are sensitivity, DB index. The calculations of metrics are as follows:

$$Sensitivity = \frac{number\ of\ true\ positives}{number\ of\ true\ positives + false\ negatives}$$

DB Index

The Davies Bouldin (DB) Index is a metric exploited to evaluate the clustering algorithm. The DB-Index is an internal evaluation scheme that validates how well the cluster is done based on the quantities and features inherent to the dataset. The DB-Index calculation is as follows:

$$DBI = \frac{1}{N}\sum_{n=1}^{N} D_{n,n+1}$$

Where, $D_{n,n+1} = \dfrac{d_n + d_{n+1}}{M}$

$$d_n = \frac{1}{T}\sum_{b=1}^{T}\left|X_b - C_n\right|^2$$

$$M = \sum_{n=1}^{N-1}\sum_{f=n+1}^{N}\sqrt{\left(C_n - C_f\right)^2}$$

In the above equations DBI denotes the Davies Bouldin (DB) Index, N denotes total number of clusters, $d_{n,n+1}$ denotes clustering scheme measurement between each cluster, dn denotes the value of distance between each data in the n^{th} cluster and centroid of that cluster, dn+1 denotes the value of distance between each data in the next cluster and the centroid of n^{th} cluster, M denotes sum of the Euclidean distance between each centroid, T is the total number of data in the cluster, X is the data in the n^{th} cluster and C_n is the centroid of n^{th} cluster.

Performance Comparison

The performance of proposed technique is compared with the existing segmentation algorithms such as ABC, KFCM in terms of external metrics and internal metrics using satellite images. The Fig.5(a) shows the satellite images taken for experimentation and the Fig.5(b) shows the classified regions such as grass, road, building, tree and shadow in the image.

Fig.5(a) Satellite Image

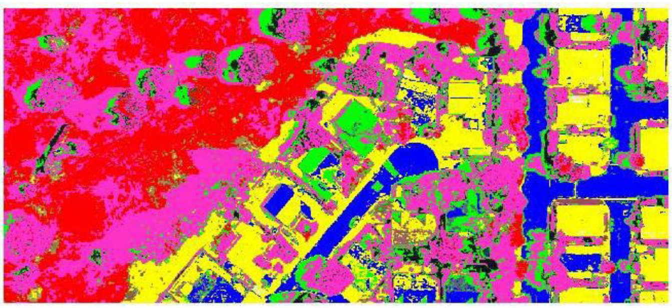

Fig.5(b) Classified regions

B .Performance Based on External Metrics

The table shows the accuracy obtained for our technique compared to the existing techniques using image taken for experimentation.

In table the sensitivity obtained for the proposed segmentation algorithm ABC-KFCM is compared with the existing techniques using taken for experimentation. As shown proposed technique performed better than all the existing algorithms taken for comparison.

Sensitivity	ABC-KFCM	ABC
Road	0.81	0.71
Building	0.85	0.80
shade	0.88	0.83
tree	0.81	0.7107
grass	0.84	0.78

Performance Based on Internal Metrics

This section shows the performance comparisons by means of DB-index for satellite image taken for experimentation. The better performances of these indices are judged based on less value. The table 1 shows the DB-index, performances of proposed technique compared to the existing techniques.

Clusteting Technique	DB-index
ABC-KFCM	0.476602
ABC	0.476602

TABLE 2

The TABLE1 shows the DB-index, XB-index and MSE comparison using image taken for experimentation.

It shows that ABC-KFCM is has very less DB-index which indicates it does better clustering.

CONCLUSION

In this paper, a new optimization algorithms for segmentation is proposed with the intention of improving the segmentation in satellite images using feed-forward neural network classifier.. The overall steps involved in the proposed technique in three steps such as, i) Pre-processing, ii) segmentation using ABC-KFCM algorithm, and iii) classification using feed-forward neural network classifier.

Classification accuracy of the proposed algorithm in satellite image classification is calculated and the performance is compared with various clustering algorithms.

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AN IMPROVED ARCHITECTURE OF 256 BIT CSLAFOR REDUCED AREA APPLICATIONS

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Abstract—In the design of Integrated Circuits, area occupancy plays a vital role because of increasing the necessity of portable systems and to reduce power consumption. In designing a adder carry generation is critical way to reduce number of transistors and to reduce power consumption. Carry select adder (CSLA) is one of the fastest adders used and there is further scope of reducing the area in CSLA structure. The proposed design is implemented by sharing the common Boolean logic term (CBL) to develop an area-efficient CSLA. one XOR and one inverter in each summation operation as well as one AND gate and one inverter in each carry-out operation and Through the multiplexer, the correct output is selected according to the logic states of the carry in signal. this paper proposes efficient SQRT CSLA (CBL CSLA) architecture in terms of area. 8-bit, 16-bit, 32-bit, 64-bit 128 bit and 256 bit architectures of CBL based CSLA are designed and compared with regular CSLA and BEC based CSLA.

Keywords—**Adder, Carry select Adder (CSLA), Common Boolean Logic(CBL), Binary to Excess-1 Converter(BEC),Exclusive or(XOR).**

I. INTRODUCTION

As the scale of integration keeps growing, more and more sophisticated electronic systems are being implemented on a VLSI chip. Adders are most widely used in multipliers, DSP to execute various algorithms like FFT, FIR and IIR. As we know millions of instructions per second are performed in microprocessors. So, speed of operation is the most important constraint to be considered while designing multipliers. Due to device portability miniaturization of device should be high and power consumption should be low. Devices like Mobile, Laptops etc. require more battery backup.

So, a VLSI designer has to optimize these three parameters in a design. These constraints are very difficult to achieve so depending on demand or application some compromise between constraints has to be made. Ripple carry adders exhibits the most compact design but the slowest in speed. Whereas carry look ahead is the fastest one but consumes more area. Carry select adders act as a compromise between the two adders. In 2002, a new concept of hybrid adders is presented to speed up addition process by Wang et al. that gives hybrid carry look-ahead/carry select adders design. In 2008, low power multipliers based on new hybrid full adders.

Design of area- and power-efficient high-speed data path logics systems are one of the most substantial areas of research in VLSI system design. In digital adders, the speed of addition is limited by the time required to propagate a carry through the adder. The sum for each bit position in an elementary adder is generated sequentially only after the previous bit position has been summed and a carry propagated into the next position.

The CSLA is used in many computational systems to alleviate the problem of carry propagation delay by independently generating multiple carries and then select a carry to generate the sum. However, the CSLA is not area efficient because it uses multiple pairs of Ripple Carry Adders (RCA) to generate partial sum and carry by considering carry input ‘0’ and ‘1’, then the final sum and carry are selected by the multiplexers (MUX).

The existing modified SQRT CSLA is to use Binary to Excess-I Converter (BEC) instead of RCA with Cin=1 in the regular CSLA to achieve lower area and power consumption with slightly increase in the delay. The basic idea of the proposed architecture is that which replaces the BEC by Common Boolean Logic.

II. BASIC STRUCTURE OF REGULAR SQRT CARRY SELECT ADDER

The carry select adder comes in the category of conditional sum adder. Conditional sum adder works on some condition. Sum and carry are calculated by assuming input carry as 1 and 0 prior the input carry comes. When actual carry input arrives, the actual calculated values of sum and carry are selected using a multiplexer. The conventional carry select adder consists of K/2 bit adder for the lower half of the bits i.e. least significant bits and for the upper half i.e. most significant bits (MSB's) two K bit adders. In MSB adders one adder assumes carry input as one for performing addition and another assumes carry input as zero. The carry out calculated from the last stage i.e. least significant bit stage is used to select the actual calculated values of output carry and sum. The selection is done by using a multiplexer. This technique of dividing adder in to two stages increases the area utilization but addition operation fastens. N bit Square root CSLA with P stages, first stage adds M bits second M+1 and so on. Hence number of stages is proportional to √N and delay is proportional to √N

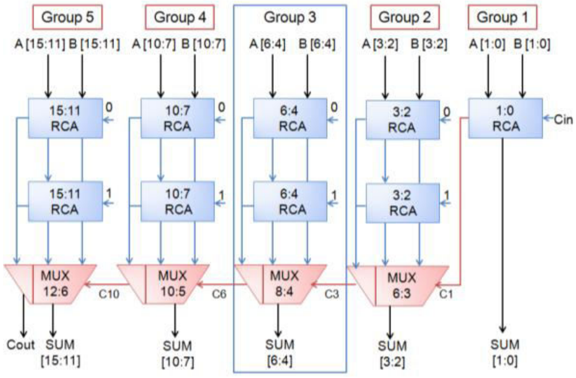
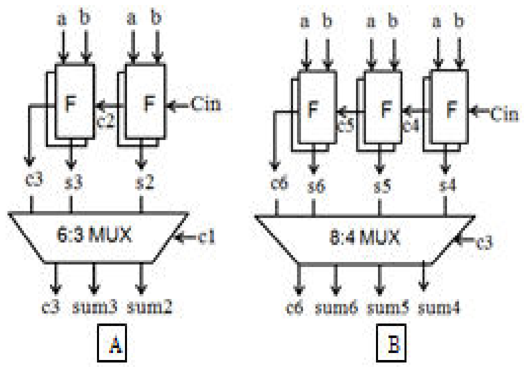


Figure 1 Block diagram of regular 16-b SQRT CSLA with groups shown.

The main disadvantage of regular CSLA is the large area due to the multiple pairs of ripple carry adder. The regular 16-bit Carry select adder is shown in Figure1 [7]. It is divided into five groups with different bit size RCA. From the structure of Regular CSLA, there is scope for reducing area and power consumption. The carry out calculated from the last stage i.e. least significant bit stage is used to select the actual calculated values of the output carry and sum. The selection is done by using a multiplexer.



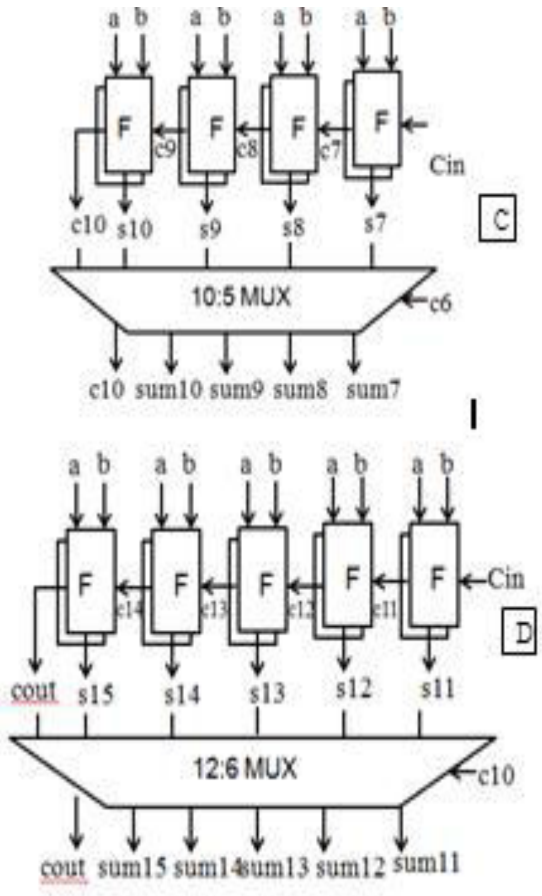


Figure2 Area evaluation methodology of 16 bit SQRT CSLA

TABLE.I

DELAY AND AREA COUNT OF THE BASIC BLOCKS OF CSLA

BLOCK	DELAY	AREA
XOR	3	5
2:1 MUX	3	4
HALF ADDER	3	6
FULL ADDER	6	13

Detailed internal structure of each group from group 2 to group 5 and the estimated maximum delay and area of the other groups in the regular SQRT CSLA

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are evaluated as shown in Figure2, Table1 and listed in Table2 and Table3.

TABLE.II

AREA COUNT CALCULATION OF INDIVIDUAL GROUPS OF 16 BIT SQRT CSLA.

GROUP	AREA CALCULATION	Area in Gates
Group 1	2 bit RCA = 2FA = 2*13 gates	26
Group 2	2bit RCA(c=0) + 2 bit RCA(c=1) + 3 bit MUX = 1FA + 1HA + 2FA + 3(2:1 MUX) = 1*13 + 1*6 + 2*13 + 3*4 gates	57
Group 3	3 bit RCA(c=0) + 3 bit RCA(c=1) + 4 bit MUX = 2FA + 1HA + 3FA + 4(2:1 MUX) = 2*13 + 1*6 + 3*13 + 4*4 gates	87
Group 4	4 bit RCA(c=0) + 4 bit RCA(c=1) + 5 bit MUX = 3FA + 1HA + 4FA + 5(2:1 MUX) = 3*13 +1*6 + 4*13 + 5*4 gates	117

TABLE III

AREA COUNT AND DELAY OF 16 BIT SQRT CSLA GROUPS

GROUP	AREA COUNT	DELAY
Group 1	26	6
Group 2	57	11
Group 3	87	13
Group 4	117	16
Group 5	147	19

III. STRUCTURE OF 16 BIT MODIFIED SQRT CSLAUSING BEC

In 16 bit SQRT CSlA with BEC, we use 3 bit BEC, 4 bit BEC, 5 bit BEC, 6 bit BEC in second row replacing RCAs with Cin = 1 . In each group for Cin=0 addition is perform using RCA. For Cin=1 addition is perform using BEC logic. From both the generated result only one output is selected based on carry in signal (actual Cin) from the previous group.Figure3. shows the block diagram of modified SQRT CSlA.

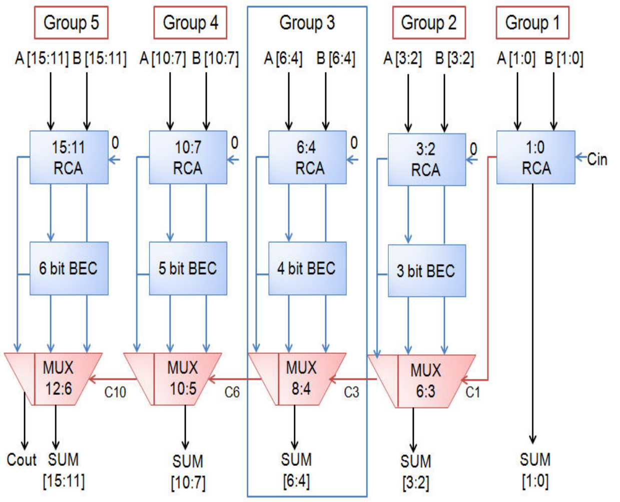

Figure 3Modified SQRT Carry Select Adder parallel RCA with Cin=1 is replaced with BEC

The delay and area estimation of each group are shown in Figure4 and Figure5 below. The steps leading to the evaluation are given here.

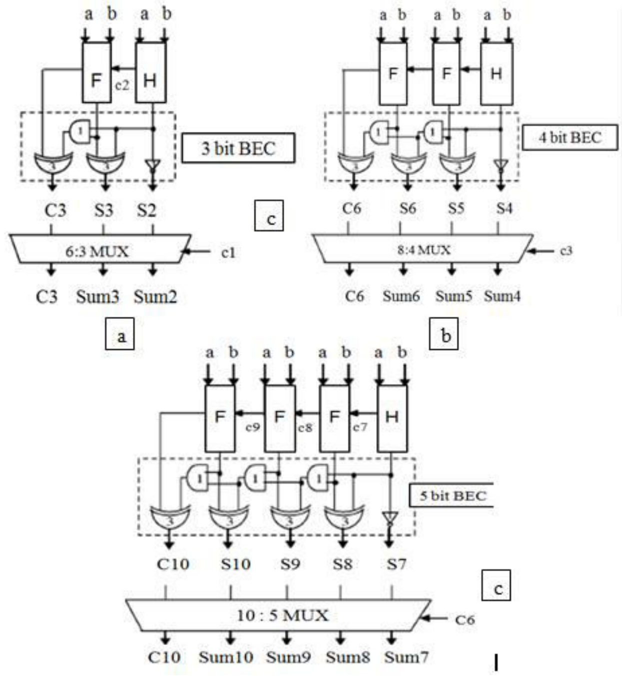

Figure 4 Delay and area evaluation of modified 16 bit SQRT CSlA: (a) Group2, (b) Group3 and (c) Group4. H is a Half Adder F is Full adder.

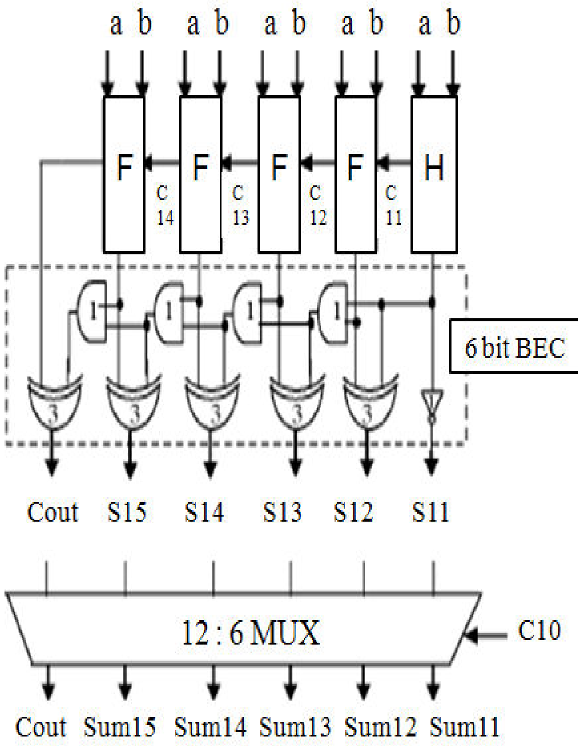

Figure 5 Delay and area evaluation of modified 16 bit SQRT CSlA: (d) Group 5H is a Half Adder F is Full adder.

The Group2 has one 2-b RCA which has 1 FA and 1 HA for Cin=0 Instead of another 2-b RCA with Cin=1 a 3-b BEC is used which adds one to the output from 2-b RCA. Based on the consideration of delay values of Table I, the arrival time of selection input C1 [t=7] of 6:3 MUX is earlier than the s3 [t=9] and C3 [t=10] and later than the S2 [t=4]. Thus, the Sum3 and final C3 (output from MUX) are depending on S3 and MUX and partial C3 (input to MUX) and MUX, respectively. The Sum2 depends on C1 and MUX.

The area count of group2 is determined as shown

Gate count = 43 (FA + HA + MUX + BEC)

FA =13 (1*13)

HA = 6 (1 * 6)

MUX =12 (3*4)

BEC (3-BIT) : NOT = 1, AND =1, XOR = 10 (2 * 5)

Similarly, the estimated maximum delay and area of the other groups of the modified SQRT CSlA are evaluated inTable4 and Table5.

TABLE IV

AREA COUNT AND DELAY OF MODIFIED 16 BIT SQRT CSLA (BEC)

GROUP	AREA COUNT	DELAY
Group 1	26	11
Group 2	43	13
Group 3	66	16
Group 4	89	19
Group 5	113	22

TABLE.V

AREA COUNT CALCULATION OF INDIVIDUAL GROUPS OF 16 BIT MODIFIED SQRT CSLA USING BEC.

GROUP	AREA CALCULATION	Area in Gates
Group 1	2 bit RCA = 2FA = 2*13 gates	26
Group 2	2bit RCA(c=0) + 3 bit BEC + 3 bit MUX = 1FA + 1HA + 1 NOT + 1AND + 2XOR + 3(2:1 MUX) =1*13 + 1*6 + 1*1 + 1*1 + 2*5 + 3*4 gates	43
Group 3	3 bit RCA(c=0) + 4 bit BEC + 4 bit MUX = 2FA + 1HA + 1 NOT + 2AND + 3XOR + 4(2:1 MUX) =2*13 + 1*6 + 1*1 + 2*1 + 3*5 + 4*4 gates	66
Group 4	4 bit RCA(c=0) + 5 bit BEC + 5 bit MUX = 3FA + 1HA + 1 NOT + 3AND + 4XOR + 5(2:1 MUX) =3*13 +1*6 + 1*1 + 3*1 + 4*5 + 5*4 gates	89
Group 5	5 bit RCA(c=0) + 6 bit BEC + 6 bit MUX = 4FA + 1HA + 1 NOT + 4AND + 5XOR + 6(2:1 MUX) =4*13 + 1*6 + 1*1 + 4*1 + 5*5 + 6*4 gates	113

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IV. COMMON BOOLEAN LOGIC

To remove the duplicate adder cells in the conventional CSlA, an area efficient SQRT CSlA is proposed by sharing Common Boolean Logic (CBL) term. While analysing the truth table of single bit full adder, results show that the output of summation signal as carry-in signal is logic “0” is inverse signal of itself as carry-in signal is logic “1”. It is illustrated by red circles in Figure6. To share the Common Boolean Logic term, we only need to implement a XOR gate and one INV gate to generate the summation pair. And to generate the carry pair, we need to implement one OR gate and one AND gate. In this way, the summation and carry circuits can be kept parallel.

CIN	A	B	S0	COUT
0	0	0	0	0
0	0	1	1	0
0	1	0	1	0
0	1	1	0	1
1	0	0	1	0
1	0	1	0	1
1	1	0	0	1
1	1	1	1	1

Figure 6Truth table of single bit full adder

Where the upper half part is the case of Cin = 0 and the lower half part is the case of Cin = 1. S0 result of lower part is inverse of S0 result of lower part i.e~(A ^ B). Whereas Cout of upper part can be implemented using AND gate and lower part with OR gate.

V. STRUCTURE OF 16 BIT PROPOSED CSLA USING CBL

This method replaces the Binary to Excess-1 converter add one circuit by common Boolean logic. As compared with modified SQRT CSlA, the proposed structure is little bit faster. Internal structure of proposed CBL logic is shown in Figure7.

In the proposed SQRT CSlA, the transistor count is trade-off with the speed in order to achieve lower power delay product. Thus the proposed SQRT CSlA using CBL is better than all the other designed adders. Figure8 shows the block diagram of Proposed SQRT CSlA..

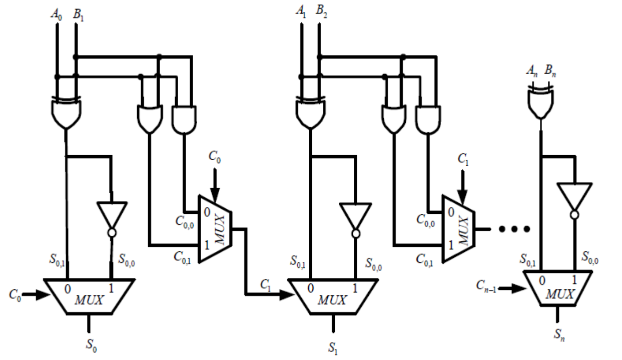


Figure7 Internal structure of the proposed area-efficient carry select adder is constructed by sharing the common Boolean logic term

In 16 bit SQRT CSlA with CBL, we use 2 bit BEC, 3 bit BEC, 4 bit BEC, 5 bit BEC in second row replacing RCAs with Cin = 1 . In each group for Cin=0 addition is perform using RCA. For Cin=1 addition is perform using CBL logic .From both the generated result only one output is selected based on carry in signal (actual Cin) from the previous group .

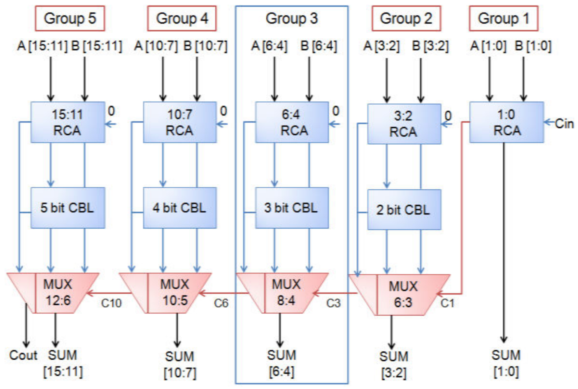


Figure8 16-Bit Proposed SQRT CSlA using Common Boolean Logic

The structure of the 16-bit Proposed SQRT CSlA is shown in Figure8It has 5 groups of different size RCA and CBL. Each group contains one RCA, one CBL and MUX. In the Proposed SQRT CSlA, The group3 performed a three bit addition which are A[4] with B[4], A[5] with B[5] and A[6] with B[6]. This is done by one half adders (HA) and two full adders (FA). The CBL block has a 4:2 multiplexer to select the appropriate carryout and summation signal for carry-in signal “1”. Through 2:1 multiplexer the carry signal is propagate to the next adder cell. The

6:3 MUX and 4:2 MUX is the combination of 2:1 MUX.

If the C3 = 0, The MUX select RCA output otherwise it select CBL output. The output of Group3 are Sum [6:4] and carryout, C6. Then the area count of Group3 is determined as shown.

Gate count = 54 (FA + HA + MUX + CBL)

FA =26 (2*13)

HA = 6 (1 * 6)

MUX =16 (4*4)

CBL (3-BIT) = 3NOT + 3OR = 6

Similarly the estimated area of the other groups in the modified SQRT CSlAshown in Figure9and Figure10are evaluated and listed in Table 6

TABLE VI

AREA COUNT AND DELAY OF PROPOSED 16 BIT SQRT CSLA

GROUP	AREA COUNT
Group 1	26
Group 2	35
Group 3	54
Group 4	73
Group 5	92

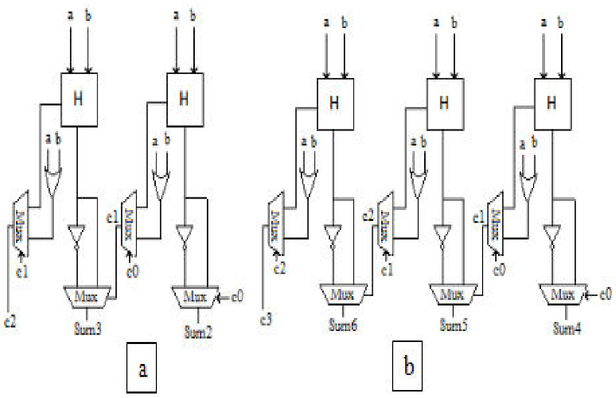


Figure9Area evaluation of proposed SQRT CSlA using CBL: (a) group2.(b) Group3. H is a Half Adder

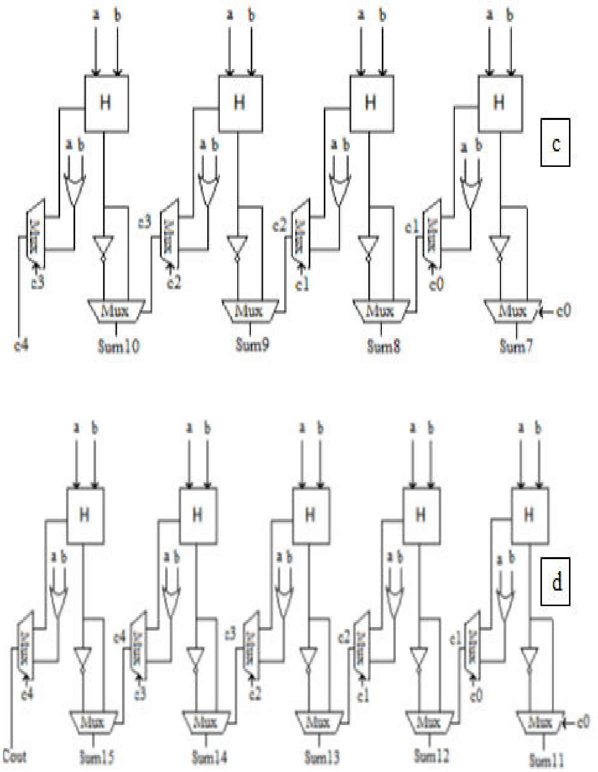


Figure10Area evaluation of proposed SQRT CSlA using CBL:

(c) Group4 and (d) group5.H is a Half Adder

VI. RESULTS

This work has been developed using Xilinx 9.2i tool. Table 8.1 shows the comparison between the various adders like conventional regular SQRT CSlA, modified SQRT CSlA (BEC) and proposed SQRT CSlA (CBL) for 8-bit, 16-bit, 32-bit, 64-bit, 128-bit and 256-bit. Comparison depicts that the proposed SQRT CSlA has less number of gates and hence less area. It is clear that area, power of proposed SQRT CSlA is reduced as compared to other adders

In following simulation results a, b are N bit data, Cin is input carry signal, Cout is carry out signal and's' is N bit summation result of a and b.

Inputs:

a[255:0] = 256'h5f52356566489afdcd456489456,

b[255:0]=256'haaaa58791234bc4587d986ca123564647, cin=0

Outputs:

s[255:0]=256'haaaa58d8646a21abd07484975799eda9d , cout = 0

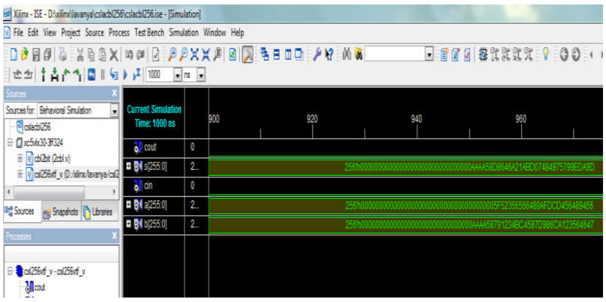
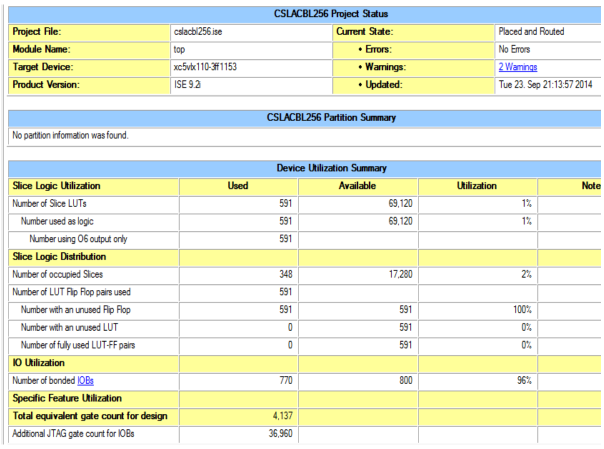


Figure 11Simulation result of 256 bit Regular, BEC, CBL SQRT CSlA

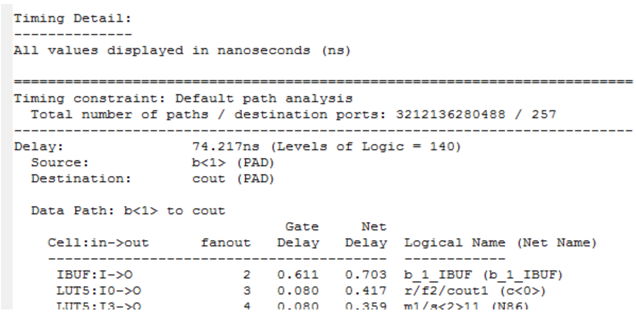


CSLACBL256 Project Status			
Project File:	cslacbl256.ise	Current State:	Placed and Routed
Module Name:	top	• Errors:	No Errors
Target Device:	xc5vlx110-3ff1153	• Warnings:	2 Warnings
Product Version:	ISE 9.2i	• Updated:	Tue 23. Sep 21:13:57 2014

CSLACBL256 Partition Summary
No partition information was found.

Device Utilization Summary				
Slice Logic Utilization	Used	Available	Utilization	Note
Number of Slice LUTs	591	69,120	1%	
Number used as logic	591	69,120	1%	
Number using O6 output only	591			
Slice Logic Distribution				
Number of occupied Slices	348	17,280	2%	
Number of LUT Flip Flop pairs used	591			
Number with an unused Flip Flop	591	591	100%	
Number with an unused LUT	0	591	0%	
Number of fully used LUT-FF pairs	0	591	0%	
IO Utilization				
Number of bonded IOBs	770	800	96%	
Specific Feature Utilization				
Total equivalent gate count for design	4,137			
Additional JTAG gate count for IOBs	36,960			

Figure 12Device utilization summary of 256 bit CBL SQRT CSlA

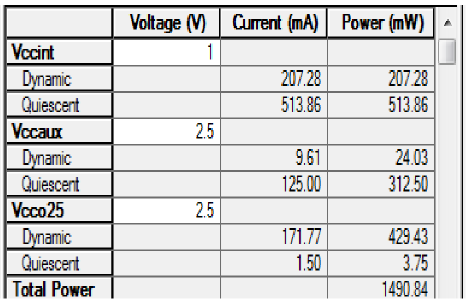


```
Timing Detail:
--------------
All values displayed in nanoseconds (ns)

=========================================================================
Timing constraint: Default path analysis
  Total number of paths / destination ports: 3212136280488 / 257
-------------------------------------------------------------------------
Delay:               74.217ns (Levels of Logic = 140)
  Source:            b<1> (PAD)
  Destination:       cout (PAD)

  Data Path: b<1> to cout
                                Gate     Net
    Cell:in->out      fanout   Delay   Delay  Logical Name (Net Name)
    ----------------------------------------  ------------
     IBUF:I->O             2   0.611   0.703  b_1_IBUF (b_1_IBUF)
     LUT5:I0->O            3   0.080   0.417  r/f2/cout1 (c<0>)
     LUT5:I3->O            4   0.080   0.359  m1/s<2>11 (N86)
```

Figure 13 Timing details of 256 bit CBL SQRT CSlA



	Voltage (V)	Current (mA)	Power (mW)
Vccint	1		
Dynamic		207.28	207.28
Quiescent		513.86	513.86
Vccaux	2.5		
Dynamic		9.61	24.03
Quiescent		125.00	312.50
Vcco25	2.5		
Dynamic		171.77	429.43
Quiescent		1.50	3.75
Total Power			1490.84

Figure 14Power analyser report of 256 bit CBL SQRT CSlA

VII. COMPARISONS

This work has been developed using Xilinx 9.2i tool. Table 7 shows the comparison between the various adders like conventional regular SQRT CSlA, modified SQRT CSlA (BEC) and proposed SQRT CSlA (CBL) for 8-bit, 16-bit, 32-bit, 64-bit, 128-bit and 256-bit. Each of the CSlA implementation is done individually using higher bit BEC and CBL logic. Inputs are directly given to the adder to complete the sum and carry.

Table 8 exhibits the delay; area of regular, modified and proposed CSlA Simulation is carried out using Xilinx 9.2i simulation tool and Vertex 5 as the target device. Comparison depicts that the proposed SQRT CSlA using CBL logic has less number of gates than the regular and BEC based CSlA with slight increase in the delay and hence less area. Reduction in area is 10% to 20% for different number of bits and reduction in area gradually decreasing with higher bits. It is clear that area, power of proposed SQRT CSlA is reduced as compared to other adders. Table 8.1 shows comparison of Results of proposed design without cascading and results of previous design obtained by cascading Spartan 3e as target device.

TABLE VII

COMPARISON OF 8, 16, 32, 64 BITS REGULAR, BEC CBL SQRT CSLAS USING SPARTAN 3E

No of bits		Total equivalent gate count		Delay (ns)	
		Previous [1]	Proposed	Previous [1]	Proposed
8	Reg	144	117	11.92	12.085
	BEC	132	96	13.69	13.125
	CBL	111	120	11.15	11.143
16	Reg	348	264	16.15	16.710
	BEC	291	261	18.77	15.361
	CBL	276	234	15.48	19.260
32	Reg	698	582	28.97	24.454
	BEC	762	609	34.44	19.969
	CBL	552	495	26.23	30.883
64	Reg	1592	1266	52.82	36.735
	BEC	1498	1275	64.61	27.410
	CBL	1104	1191	47.4	59.255

TABLE VIII

AREA AND DELAY COMPARISON OF 8 BIT,16 BIT,32BIT,64 BIT,128 BIT,256 BIT REGULAR, BEC,CBL,SQRT CSLAS USING VERTEX 5

No of bits		Total equivalent gate count	Delay (ns)
8	Reg	105	4.871
	BEC	91	5.547
	CBL	91	5.727
16	Reg	224	6.379
	BEC	231	7.709
	CBL	196	6.917
32	Reg	490	8.087
	BEC	532	10.118
	CBL	427	10.940
64	Reg	1029	10.628
	BEC	1092	11.219
	CBL	945	18.764
128	Reg	2086	14.689
	BEC	2261	14.723
	CBL	1953	34.048
256	Reg	4200	23.976
	BEC	4907	31.686
	CBL	4137	74.217

V. CONCLUSION

This paper has really given an effective description of a higher bit area efficient carry select adder. This has been achieved by altering the logic blocks of the regular module, By sharing the common Boolean logic (CBL) term, the duplicated adder cells in the conventional carry select adder is removed. A regular CSLA uses two copies of the carry evaluation blocks, one with block carry input is zero and other one with block carry input is one. The Regular SQRT CSLA has the disadvantage of more power consumptions and occupying more chip area.The reduced number of gates of this work offers the great advantage in the reduction of area. This paper proposes a scheme for 128 bit and 256 bit which reduces the area than the regular and modified SQRT CSLA 10% to 20%. Equivalent gate count of proposed SQRT CSLA is reduced having less area and low power which makes it simple and efficient for VLSI hardware implementations.

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MODIFIED BAYESIAN SPECTRUM SENSING APPROACH TO DETECT PRIMARY USER IN COGNITIVE RADIO

[1] S. RAMYA KRISHNA, [2] S. ARUNA KUMARI,

Abstract- In cellular networks, allocating spectrum without any interference is the major interference. So to allocate spectrum to secondary user without any interference with the primary user, secondary user must perform spectrum sensing before attempting to transmit over channel. This can be achieved by cognitive radios. By using cognitive radio, secondary user can use the channel or spectrum allocated to primary users as long as user not using.

In this paper a modified Bayesian detector is implemented to detect the primary user in the spectrum. The performance of the modified Bayesian detector is compared with the energy detector, Bayesian detector in high signal to noise ratio and in low signal to noise ratio. The performance analysis of detector is analyzed using probability of detection and probability of false alarm. MATLAB simulation environment is used for the performance estimation.

Keywords- Modified Bayesian; cognitive radio; primary user; secondary user; SNR; energy detector; false alarm; detection.

I. INTRODUCTION

The available electromagnetic radio spectrum is a limited natural resource and getting crowded day by day due to increase in wireless devices and applications. It has been also found that the allocated spectrum is underutilized because of the static allocation of the spectrum. Also, the conventional approach to spectrum management is very inflexible in the sense that each wireless operator is assigned an exclusive license to operate in a certain frequency band. And, with most of the useful radio spectrum already allocated, it is difficult to find vacant bands to either deploy new services or to enhance existing ones. In order to overcome this situation, we need to come up with a means for improved utilization of the spectrum creating opportunities for dynamic spectrum access. [1] - [3].

The issue of spectrum underutilization in wireless communication can be solved in a better way using Cognitive radio (CR) technology. The term 'cognitive radio' is given to an emerging wireless access scheme that is 'an intelligent wireless communication system that is aware of its surrounding environment to allow changes in certain operating parameters for the objective of providing reliable communication and efficient utilization of the radio spectrum' [4].

A major challenge in cognitive radio is that the secondary users need to detect the presence of primary users in a licensed spectrum and quit the frequency band as quickly as possible if the corresponding primary radio emerges in order to avoid interference to primary users. This technique is called spectrum sensing. Spectrum sensing and estimation is the first step to implement Cognitive Radio system [5]. We can categorize spectrum sensing techniques into direct method, which is considered as frequency domain approach, where the estimation is carried out directly from signal and indirect method, which is known as time domain approach, where the estimation is performed using autocorrelation of the signal. Another way of categorizing the spectrum sensing and estimation methods is by making group into model based parametric method and periodogram based nonparametric method.

Organization of this paper is as followed, Section 2 deal with conventional spectrum sensing detectors. Section 3 deal with modified Bayesian detector implementation and section 4 deal with the simulation results comparison and its analysis. Section 5 concludes the paper.

II. CONVENTIONAL DETECTORS

A. Energy Detector.

Energy detector [7] is the most popular way of spectrum sensing because of its low computational and implementation complexities. The receivers do not need any knowledge about the primary users. An energy detector (ED) simply treats the primary signal as noise and decides on the presence or absence of the primary signal based on the energy of the observed signal.

A decision statistic for energy detector is:

$$T = \sum_{N} (Y[n])^2$$

$T > \gamma$ *decide signal present*

$T < \gamma$ *decide signal absent*

Where γ is threshold.

B. Matched Filter

The matched filter (also referred to as coherent detector), it can consider as a best sensing technique if CR has knowledge of PU. It is very accurate because it maximizes the received signal-to-noise ratio (SNR). Matched filter coefficients are basically given by the complex conjugated reversed signal samples in terms of discrete signals. Two types of coherent or non-coherent receivers are used based on signal analysis either as complex signals or noises. If the amplitude and phase of the received signal are known coherent receivers are used results in a perfect match between the matched filter coefficients and the signals. In case of a non-coherent receiver, the received signal is modeled as a replica of the original signal with a random phase error. With a non-coherent receiver the detection after the matched filter is generally based on the power or magnitude of the signal since we need both real and imaginary parts to define the signal entirely [6].

C. Cyclostationary Detection

Implementation of a Cyclostationary feature detector is a spectrum sensing which can differentiate the modulated signal from the additive noise. A signal is said to be Cyclostationary if its mean and autocorrelation are a periodic function. Cyclostationary feature detection can distinguish PU signal from noise and used at very low Signal to Noise Ratio (SNR) detection by using the information present in the PU signal that is not present in the noise.

III. MODIFIED BAYESIAN APPROACH

We study the approximation of our proposed detector for MPSK modulated primary signals in the low SNR regime. Through approximation, the detector structure becomes:

$$\frac{1}{N}\sum_{k=0}^{N-1}\sum_{n=0}^{\frac{M}{2}-1}\left(\Re\left[r(k)h^* e^{-j\phi_n(k)}\right]\right)^2 \underset{\mathcal{H}_0}{\overset{\mathcal{H}_1}{\gtrless}} \frac{MN_0^2}{4}\left(\gamma + \frac{\ln \epsilon}{N}\right)$$

The proposed detector is an energy detector in the low SNR regime for MPSK signals (M> 2).The detector can be normalized to

$$T_{\text{L-ABD-1}} = \frac{1}{N}\sum_{k=0}^{N-1}|r(k)|^2 \underset{\mathcal{H}_0}{\overset{\mathcal{H}_1}{\gtrless}} \frac{N_0}{\gamma}\left(\gamma + \frac{\ln \epsilon}{N}\right)$$

When the signal is BPSK, the detector is equivalent to

$$T_{\text{L-ABD-1}} = \frac{1}{N|h|^2}\sum_{k=0}^{N-1}(\Re[r(k)h^*])^2 \underset{\mathcal{H}_0}{\overset{\mathcal{H}_1}{\gtrless}} \frac{N_0}{2\gamma}\left(\gamma + \frac{\ln \epsilon}{N}\right)$$

Through approximation in the high SNR regime, the detector structure (H-ABD) becomes

$$T_{\text{H-ABD}} = \frac{1}{N}\sum_{k=0}^{N-1}\ln\left(\sum_{n=0}^{M/2-1} e^{\frac{2}{N_0}\Re\left[r(k)h^* e^{-j\phi_n(k)}\right]}\right) \underset{\mathcal{H}_0}{\overset{\mathcal{H}_1}{\gtrless}} \gamma + \ln M + \frac{\ln \epsilon}{N}.$$

The suboptimal BD detector employs the sum of received signal magnitudes to detect the presence of primary signals in the high SNR regime, which indicates that energy detector is not optimal in this regime. Similar to the derivation we can derive the suboptimal detector as shown in which also uses the sum of the real part of the received signal magnitudes to detect primary signals. The detector H- ABD is as follows:

$$T_{\text{H-ABD}} = \frac{1}{N}\sum_{k=0}^{N-1}|\Re[r(k)h^*]| \underset{\mathcal{H}_0}{\overset{\mathcal{H}_1}{\gtrless}} \frac{N_0}{2}\left(\gamma + \ln 2 + \frac{\ln \epsilon}{N}\right)$$

IV. SIMULATION RESULTS

We assume a range for P_f and P_D for which the range of SNR values are calculated. In this simulation we used 1000 samples and BPSK modulation. The figure 1 shows the simulation plot for detection and false alarm both theoretical and practical values for the energy detection based spectrum sensing.

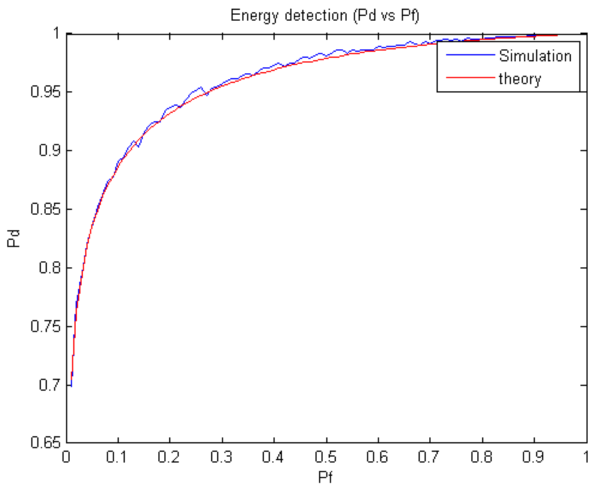

Figure.1 detection vs false alarm plot for energy detector spectrum sensing

Figure 2,3 gives the performance estimation of spectrum sensing for energy detection based and Bayesian based spectrum sensing for low SNR using BPSK modulation for 1000 samples.

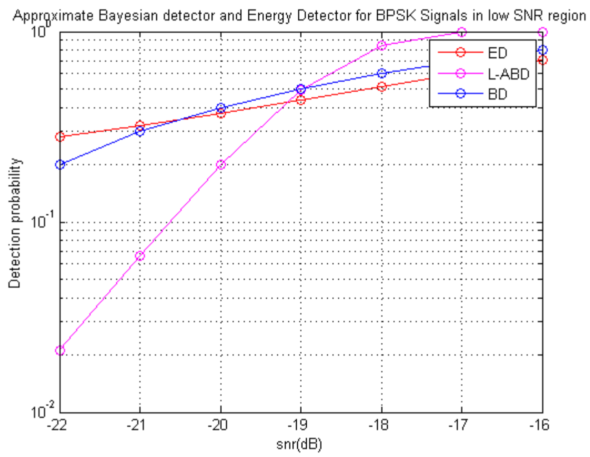

Figure 2 probability of detection for low SNR

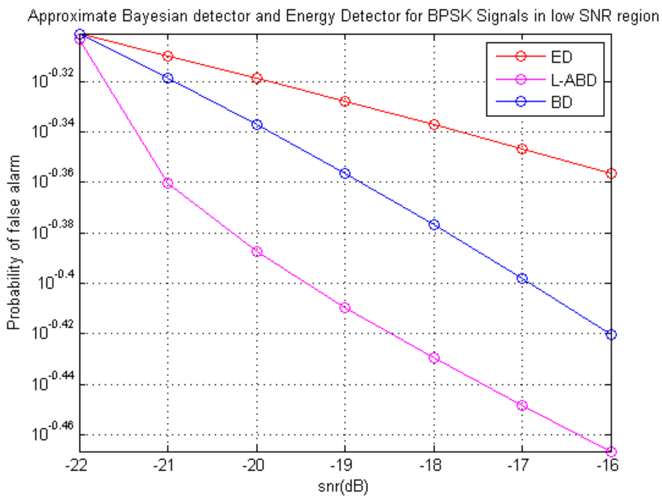

Figure.3 probability of false alarm

Figure 4, 5 gives the performance estimation of spectrum sensing for Energy detection based and Bayesian based spectrum sensing for high SNR using BPSK modulation for 1000 samples.

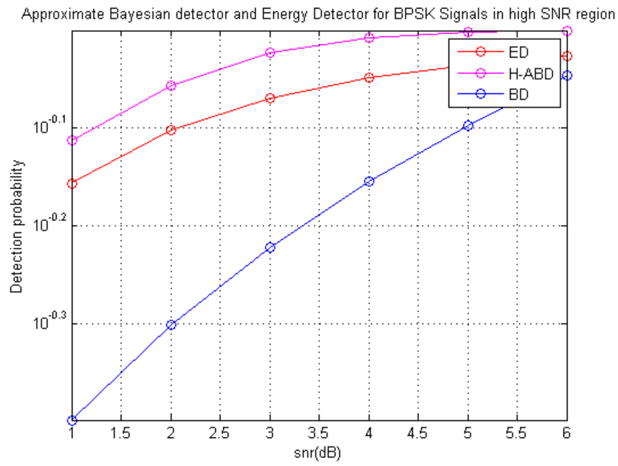

Figure 4 probability of detection for high SNR

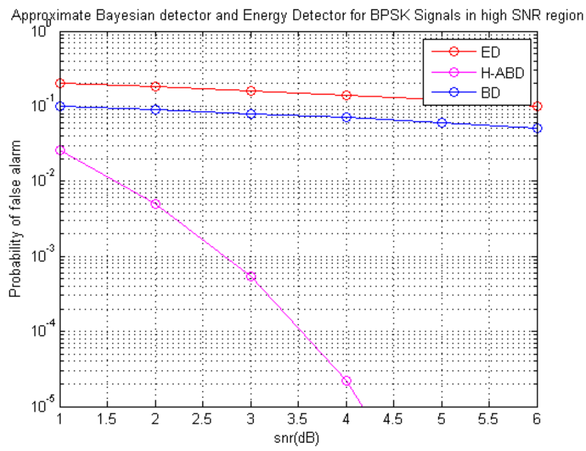

Figure.5 probability of false alarm for high SNR

Figures 6 and 7 gives the performance estimate of the system under lower SNR using M-PSK modulation. In the below results M=8 is used.

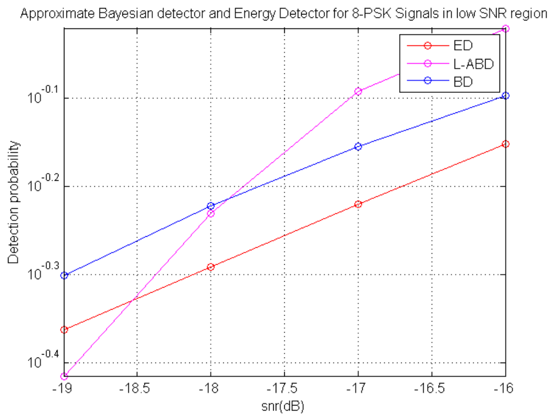

Figure 6 probability of detection for low SNR with M=8

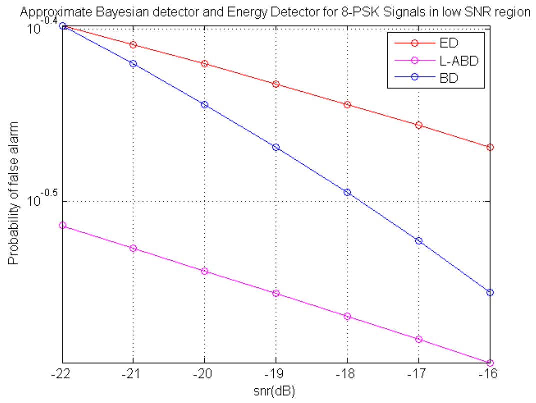

Figure.7 probability of false alarm for low SNR with M=8

Figures 8 and 9 gives the performance estimate of the system under high SNR using M-PSK modulation. In the below results M=8 is used.

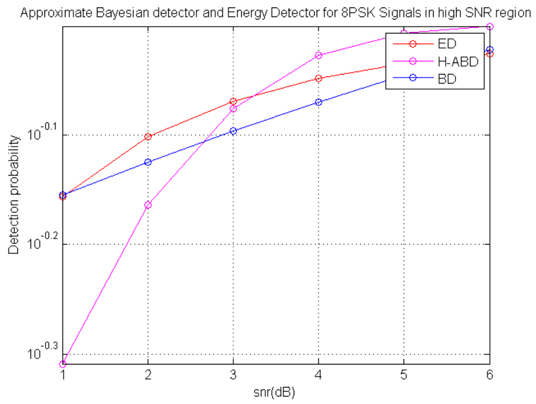

Figure 8 probability of detection for high SNR with M=8

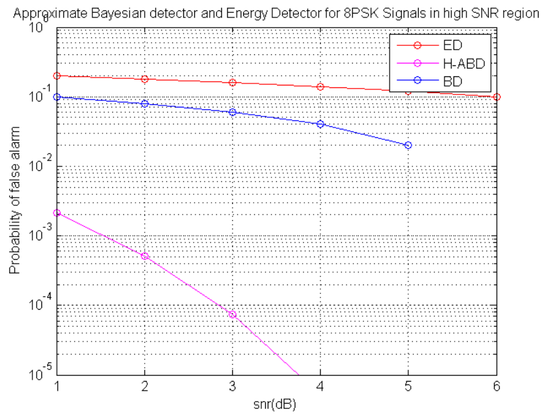

Figure.9 probability of false alarm for high SNR with M=8

V. CONCLUSION

The performance of the modified Bayesian detector is compared with the energy detector, Bayesian detector in high signal to noise ratio and in low signal to noise ratio using probability of detection and probability of false alarm in MATLAB environment.

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DESIGN AND IMPLEMENTATION OF NETWORK BASED SECURITY SYSTEM

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***Abstract- In the Present days* Security is an important concern everywhere. Industrial security keeps on getting necessary now a days as the possibilities of intrusion are increasing day by day. Safety from theft, leaking of raw gas and fire are the most important requirements of home security system for people. A traditional home security system gives the signals in terms of alarm. However, the GSM (Global System for Mobile communications) based security systems provides enhanced security as whenever a signal from sensor occurs, a text message is sent to a desired number to take necessary actions. This paper suggests a novel method for industrial security system. The system sends SMS by using microcontroller, sensors, relays and buzzers.**

Keywords- ARM7, sensor, Home Security, GSM Modem.

I. INTRODUCTION

The use of wireless communications in electronics and domestic use based commercial products is increasing day by day. These days there has been lot of work and research in the field of designing automation products for home. This has motivated us to use GSM communication network in order to achieve security in homes and intimate the owner of the home if any emergency situation has arises at industries.

Home security automation is an area of home automation. It concentrates on the safety aspects of homes and offices. It enables remote surveillance of homes and offices to improve the security of home and offices using information technology and wireless communication. The popularity of home security automation has been increased in greatly in recent years due to much higher affordability and simplicity. Security systems are valuable tools in protection against many of the dangers that can happen in and around a home. Thesesystems safeguard homes from intruders and burglars. Home security begins with home safety. Home safety begins with homeowners taking steps to protect their home and its residents.Liquid Petroleum Gas (LPG) is the mixture of propane and butane which is highly flammable chemical. It is odourless gas due to which Ethane-oil is added as powerful odorant, so that leakage can be easily detected. hazardous gas leaks by sensors. Here we intend to use a microcontroller based system where a gas sensor, MQ6 used to detect different combustible gases at low cost. This sensor has good sensitivity combined with a quick response time. ARM Microcontroller used to alert when the levels of gas detected is beyond safety limit.

II LITERATURE REVIEW

SunithaaJet al[1]The system of a wireless LPG leakage monitoring system is presented for home safety. The system detects the leakage of the LPG and alerts the user about the leak and as an emergency step the system will activate the exhaust fan and additionally checks the leakage. An additional feature of the system is that the approximate consumption is indicated in terms of the total weight. The presented system makes use of GSM module so as to alert about the gas leakage via an SMS. Whenever the system detects the rise in the concentration of the LPG it immediately alerts by activating an alarm and at the same time by sendingmessage to the desired mobile phones. The exhaust fan is switched on and an LPG safe solenoid valve fitted to the cylinder is given a signal to close avoiding further leakage. The device ensures safety and prevents suffocation and explosion due to gas leakage.

III PROPOSED MODEL

The system mainly consists of three parts. First part contains LPG gas leakage detection module. The second part the hardware is installed with proximity sensors placed at the entrance of the door to detect theft or burglaries. The third part is the ARM processor along with the GSM module. The ARM processor finds wide application due to its features and low power. The block diagram of the proposed system is shown in figure 1.Figure 2 shows the pin diagrams (or pictures) of components used.

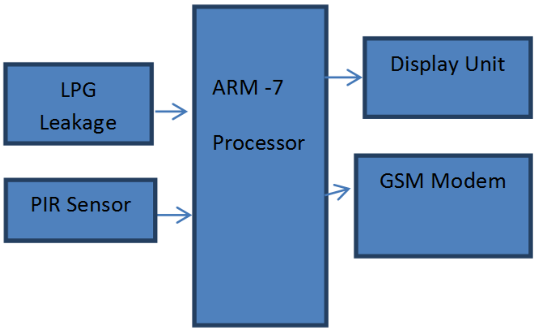

Fig 1 Block Diagram of ARM Processor

Fig 2 Components Used in ARM

A. Microcontroller LPC2148

The LPC2148 microcontrollers are based on a 32-bit ARM7TDMI-S CPU with real-time emulation and embedded trace support, which combine the microcontroller with embedded high-speed flash memory ranging from 32 kB to 512 kB. A 128-bit wide memory interface and unique accelerator architecture enable 32-bit code execution at the maximum clock rate. For critical code size applications, the alternative 16-bit Thumb mode reduces code by more than 30 % with minimal performance penalty. Due to their tiny size and low power consumption, LPC2148 are ideal for applications where miniaturization is a key requirement, such as access control and point-of-sale. Serial communications interfaces ranging from a USB 2.0 Full-speed device, multiple UARTs, SPI, SSP to I2C-bus and on-chip SRAM of 40 kB, make these devices very well suited for communication gateways and protocol converters, soft modems, voice recognition and low end imaging, providing both large buffer size and high processing power. Various 32-bit timers, single or dual 10-bit ADC(s), 10-bit DAC, PWM channels and 45 fast GPIO lineswith up to nine edge or level sensitive external interrupt pins make these microcontrollers suitable for industrial control and medical systems.

B. GSM Module

Here, a GSM modem is connected with the microcontroller. This allows the computer to use the GSM modem to communicate over the mobile network. These GSM modems are most frequently used to provide mobile Internet connectivity, many of them can also be used for sending and receiving SMS and MMS messages. GSM modem must support an “extended AT command set” for sending/receiving SMS messages. GSM modems are a cost effective solution for receiving SMS messages, because the sender is paying for the message delivery. SIM 300 is designed for global market and it is a tri-band GSM engine. It works on frequencies EGSM 900 MHz, DCS 1800 MHz and PCS 1900 MHz. SIM300 features GPRS multi-slot class 10/ class 8 (optional) and supports the GPRS coding schemes. This GSM modem is a highly flexible plug and play quad band GSM modem, interface to RS232, it supports features like voice, data, SMS, GPRS and integrated TCP/IP stack. It is controlled via AT commands (GSM 07.07,07.05 and enhanced AT commands). It uses AC – DC power adaptor with following ratings DC Voltage: 12V/1A.

C. LPG Sensor

These sensors are used in gas leakage detecting equipment in family and industry, are suitable for detecting of LPG, iso-butane, propane, LNG, avoid the noise of alcohol and cooking fumes and cigarette smoke. The sensor is composed of micro AL2O3 ceramic tube, Tin Dioxide (SnO2) sensitive layer, measuring electrode and heater are fixed into a crust made by plastic and stainless steel net. The heater provides necessary work conditions for work of sensitive components.

D. PIR Sensors

A passive infrared sensor (PIR sensor) is an electronic sensor that measures infrared (IR) light radiating from objects in its field of view. They are most often used in PIR-based motion detectors.All objects with a temperature above absolute zero emit heat energy in the form of radiation. Usually this radiation isn't visible to the human eye because it radiates at infrared wavelengths, but it can be detected by electronic devices designed for such a purpose.

The term passive in this instance refers to the fact that PIR devices do not generate or radiate any energy for detection purposes. They work entirely by detecting the energy given off by other objects.PIR sensors don't detect or measure "heat"; instead they detect the infrared radiation emitted or reflected from an object.

An individual PIR sensor detects changes in the amount of infrared radiation impinging upon it, which varies depending on the temperature and surface characteristics of the objects in front of the sensor. When an object, such as a human, passes in front of the background, such as a wall, the temperature at that point in the sensor's field of view will rise from room temperature to body temperature, and then back again. The sensor converts the resulting change in the incoming infrared radiation into a change in the output voltage, and this triggers the detection. Objects of similar temperature but different surface characteristics may also have a different infrared emission pattern, and thus moving them with respect to the background may trigger the detector as well.

PIRs come in many configurations for a wide variety of applications.

IV IMPLEMENTATION OF SECURITY SYSTEM:

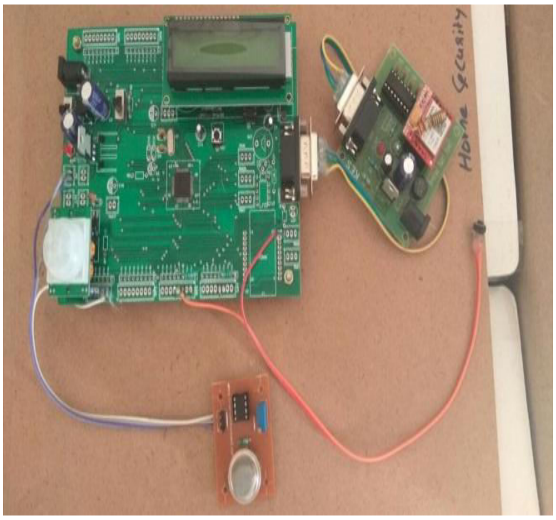

Fig 3. ARM-7 Processors

This work alters the existing security model installed in industries and this framework can also be used in homes and offices. The fundamental goal of the work is design in microcontroller based toxic gas detecting and alerting framework. The perilous gases like LPG and propane were detected and displayed each and every second in the LCD display. At the moment these gases exceed the typical level then an alarm is produced and also an alert message is sent to the approved individual through the GSM. The upside of this automatic detection and alerting system over the manual method is that it offers quick response time and precise detection of acrisis and in turn leading tofaster diffusion of the basic circumstance.

V. EXPERIMENTAL RESULTS

The prototype kit for the proposed method has been shown in the diagram given below.

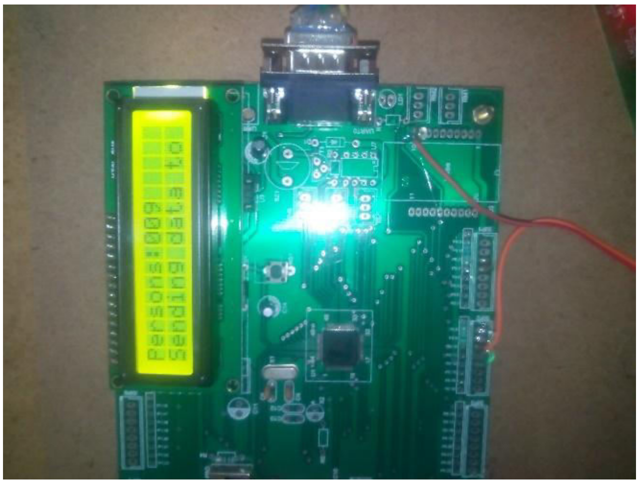

Fig 4: Output

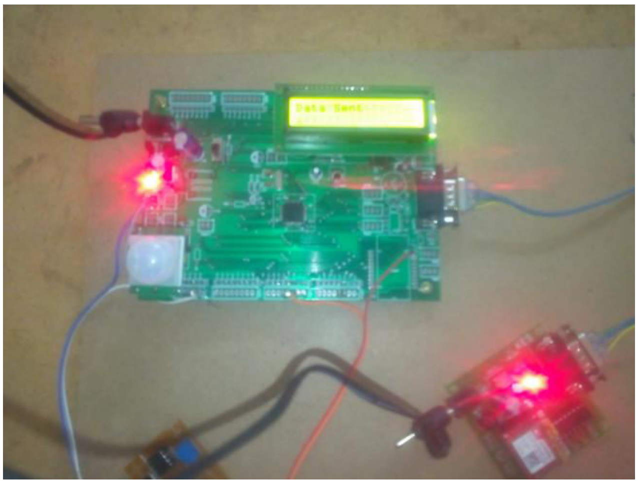

Fig 5 Output

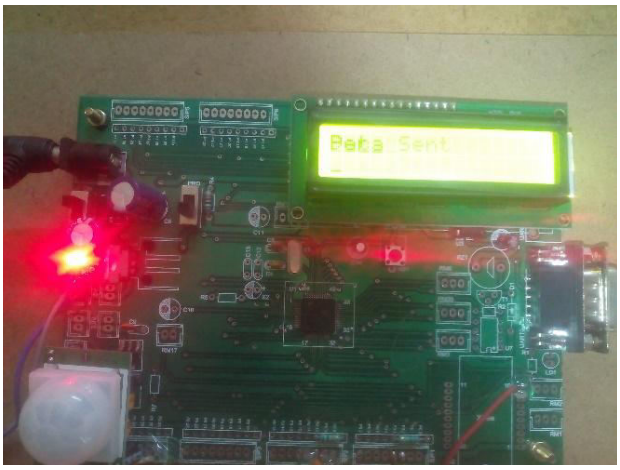

Fig 6: Output

Based on the presence of thief, or increase in LPG concentration of gas, an SMS is sent to the approved user.

VI. CONCLUSIONS AND ENHANCEMENTS

To avoid Fire acidents we implemented a new product with IR sensors. The product was tested and working properly. The main intension of the project is to prevent Fire acidents. By using this project many human lives can be saved. This project can work in any atmospheric conditions. Without any human involvement the signals will change automatically stops, if the sensors get activated in such a way that they predict a collision to occur.

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"CLOUD BASED LIGHT INTENSITY MONITORING SYSTEM USING RASPBERRY PI WITH WIRELESS DETECTION"

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ABSTRACT: **Accurate and quantifiable measurement of light is essential in creating desired outcomes in practical day to day applications as well as unique applications such as traffic lighting system, Poultry Industry, Gardening, Museum lighting system, at emergency exits etc. Hence, Light measurement and analysis is an important step in ensuring efficiency and safety. Many of the industries are burdened with limited number of resources and real shortage of experts on their fields; real time remote monitoring presents an effective solution that minimizes their efforts and expenditures to achieve the desired results within time.**

Keywords: **Light Intensity; Wireless Detection; Raspberry Pi; Cloud Storage**

I. INTRODUCTION

The Light Intensity Meter is a pocket-size, portable, light-sensitive instrument capable of reflectance and transmittance up to 10,000 foot-candles (lumens per square foot). The logarithmic response of the meter provides the operator with accurate meter deflections and easy reading at all light intensities. The Universal Light Sensor Probe, specifically designed to be maneuverable to any lighting angle or condition, can be extended to a distance of up to 2.5 feet, giving the operator a much greater degree of flexibility than a self contained instrument could do. A clear epoxy dome is affixed over the photo cell as a permanent protective feature to assure maximum durability. The "Table of Light Requirements" was compiled from materials supplied by the Department of Agriculture & other authoritative sources.

The Raspberry Pi 3 Model B is the third generation Raspberry Pi. This powerful credit-card sized single board computer can be used for many applications and supersedes the original Raspberry Pi Model B+ and Raspberry Pi 2 Model B. While maintaining the popular board format the Raspberry Pi 3 Model B brings you a more powerful processor, 10x faster than the first generation Raspberry Pi.Additionally it adds wireless LAN & Bluetooth connectivity making it the ideal solution for powerful connected designs.

There are many applications for Light Meters such as measuring and maintaining adequate light levels in schools, hospitals, production areas, laboratories, passageways and more. Adequate light levels in the work place ensure a healthier and safer environment for people. Some of important locations and light intensity is shown in TABLE

Table 1: Optimum Average Light Intensity at Various Locations

Location	Illuminance (Lux)
Warehouses, Homes, Theaters, Archives	150
Library(Reading Area)	200
Classroom	300
Laboratory	500
General office work	500

II. RELATED WORK:

A. EXISITING METHOD

There are many applications for Light meters such as measuring and maintaining adequate light levels in schools, hospitals, production areas, laboratories, passageways and more. Adequate light level in the work place ensures a healthier and safer environment for people .Some of important locations and light intensity.

It is not possible to rely upon eyesight to give accurate information about light intensity because eye adapt to changing light conditions [4].

Fig.1 Existing method

B. PROPOSED METHOD

This proposed method explains about quantifiable measurement of light intensity which are introduces real remote light intensity monitoring system using raspberry pi which enables the user to track the lighting system remotely.

In the proposed system we uses Raspberry pi, Analog to Digital convertor and Light Dependent Resistor which all deals with Software Development that is Web server, Data base and Data visualization.

We have to gather and track information from the data acquisition unit for analysis. We can do by installing MySQL

Database server in raspberry pi.

It can do the following operations

- Create a data base
- Create a table
- Load data into table

II. HARDWARE DESCRIPTION

A. LDR

A LDR (Light Dependent Resistor) is variable resistor, the resistance of the LDR is inversely proportional to the light intensity, it exhibits maximum resistance in the absence of light and minimum resistance in the presence of light.LDR produces analog output voltage with respect to incident light, The Raspberry Pi computer does not have a way to read analog inputs. It is a digital-only computer. Compared to the Arduino, AVR or PIC microcontrollers that often have 6 or more analog inputs. [5]

Fig.2 LDR

In control engineering applications, it is common to have a sensing stage (which consists of a sensor), a signal conditioning stage (where usually amplification of the signal is done) and a processing stage (normally carried out by an ADC and a micro - controller). Operational amplifiers (op-amps) are commonly employed to carry out the amplification of the signal in the signal conditioning stage.

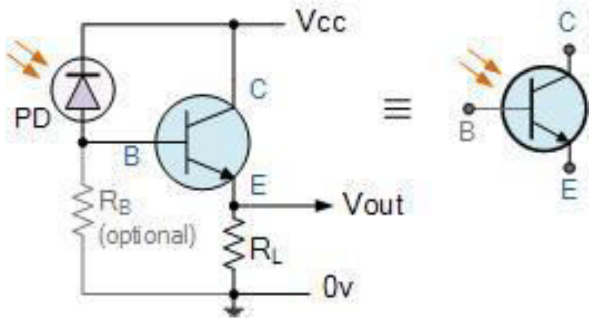


Fig.3 Light Sensors

A **Light Sensor** generates an output signal indicating the intensity of light by measuring the radiant energy that exists in a very narrow range of frequencies basically called “light”, and which ranges in frequency from “Infra-red” to “Visible” up to “Ultraviolet” light spectrum.

The light sensor is a passive device that convert this “Light energy” whether visible or in the infra-red parts of the spectrum into an electrical signal output. Light sensors are more commonly known as “Photoelectric Devices” or “Photo Sensors” because the convert light energy (photons) into electricity (electrons).

Photoelectric devices can be grouped into two main categories, those which generate electricity when illuminated, such as Photo- voltaic or Photo-emissive etc., and those which change their electrical properties in some way such as Photo-resistors or Photo-conductors. This leads to the following classification of devices.

Photo-emissive Cells – These are photo devices which release free electrons from a light sensitive material such as cesium when struck by a photon of sufficient energy. The amount of energy the photons have depends on the frequency of the light and the higher the frequency, the more energy the photons have converting light energy into electrical energy.

Photo-conductive Cells – These photo devices vary their electrical resistance when subjected to light. Photoconductivity results from light hitting a semiconductor material which controls the current flow through it. Thus, more light increase the current for a given applied voltage. The most common photoconductive material is Cadmium Sulphide used in LDR photocells.

Photo-voltaic Cells – These photo devices generate an EMF in proportion to the radiant light energy received and is similar in effect to photoconductivity. Light energy falls on to two semiconductor materials sandwiched together creating a voltage of approximately 0.5V. The most common photovoltaic material is Selenium used in solar cells.

Photo-junction Devices – These photo devices are mainly true semiconductor devices such as the photodiode or phototransistor which use light to control the flow of electrons and holes across their PN-junction. Photo junction devices are specifically designed for detector application and light penetration with their spectral response tuned to the wavelength of incident light.

A Photo conductive light sensor does not produce electricity but simply changes its physical properties when subjected to light energy. The most common type of photoconductive device is the Photo resistor which changes its electrical resistance in response to changes in the light intensity.

Photo resistors are Semiconductor devices that use light energy to control the flow of electrons, and hence the current flowing through them. The commonly used Photoconductive Cell is called the Light Dependent Resistor or LDR.

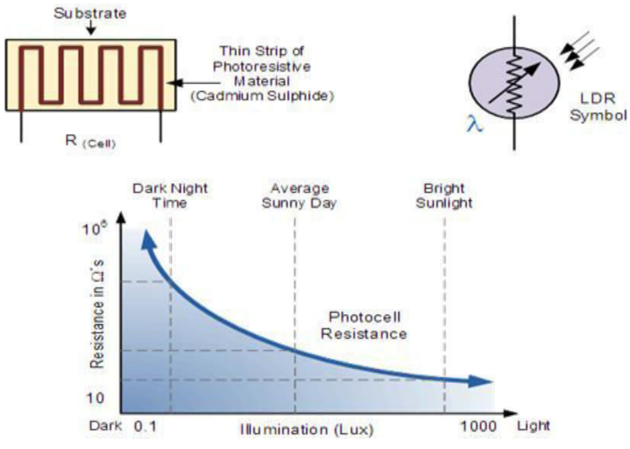


Fig.4: The Light Dependent Resistor Graph

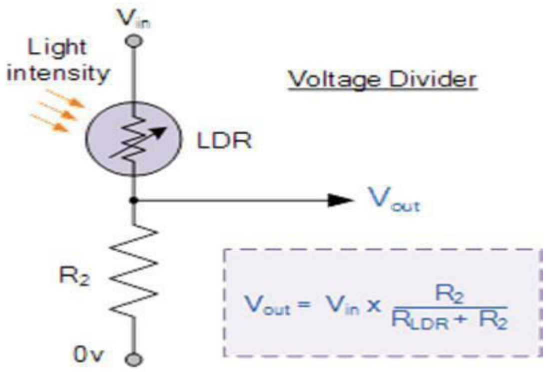


Fig.5: ORP12 Cadmium Sulphide

Photoconductive

B. ADC

Analog inputs are handy because many sensors are analog outputs, so we need a way to make the Pi analog-friendly. We can do that by wiring up an external ADC (Analog to Digital Converter) MCP 3208. The MCP 3208 acts as a bridge between digital and analog. It is a 12 bit 8 channel Analog to Digital converter. It uses the SPI bus protocol which is supported by the pi's GPIO header.

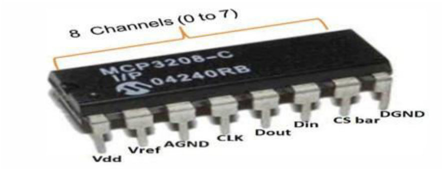


Fig.6 MCP 3208 Analog to Digital Converter

C. SIGNAL CONDITIONING CIRCUIT FOR LIGHT INTENSITY MEASUREMENT

Resistive sensors such as LDRs, RTD sand strain gages produce small percentage changes in resistance in response to a change in a physical variable such as light, temperature or force. One technique for measuring resistance is to force a constant current through the resistive sensor and measure the voltage output.

An instrumentation amplifier is a type of differential amplifier that has been outfitted with input buffer amplifiers, which eliminate the need for input impedance matching and thus make the amplifier particularly suitable for use in measurement and test equipment. Additional characteristics include very low DC offset, low drift, low noise, very high open-loop gain, very high common-mode rejection ratio, and very high input impedances. Instrumentation amplifiers are used where great accuracy and stability of the circuit both short and long-term are required.

D. Calibration of Light Dependent Resistor

The relationship between the resistance Rl and light intensity Lux for a typical LDR is

R_L =500/Lux K ohm (1)

With the LDR connected to 3.3K resistor,

The output voltage of the LDR is

$V_0 = 3.3 * R_{L/}(RL+3.3)$ (2)

From equation (1) and (2)

We obtain Light intensity Lux = ($1650/v_0$ -500)/3.3.

IV. DESIGN AND DEVELOPMENT

Complete block diagram is as shown in Fig.7 & Fig.8, signal from LDR is given to the signal conditioning circuit which is responsible to eliminate the noise, output of signal conditioning circuit given to the one of the analog channel of ADC which converts signal into digital signal, then the signal given to the GPIO (General Purpose input/output) of the Raspberry Pi.

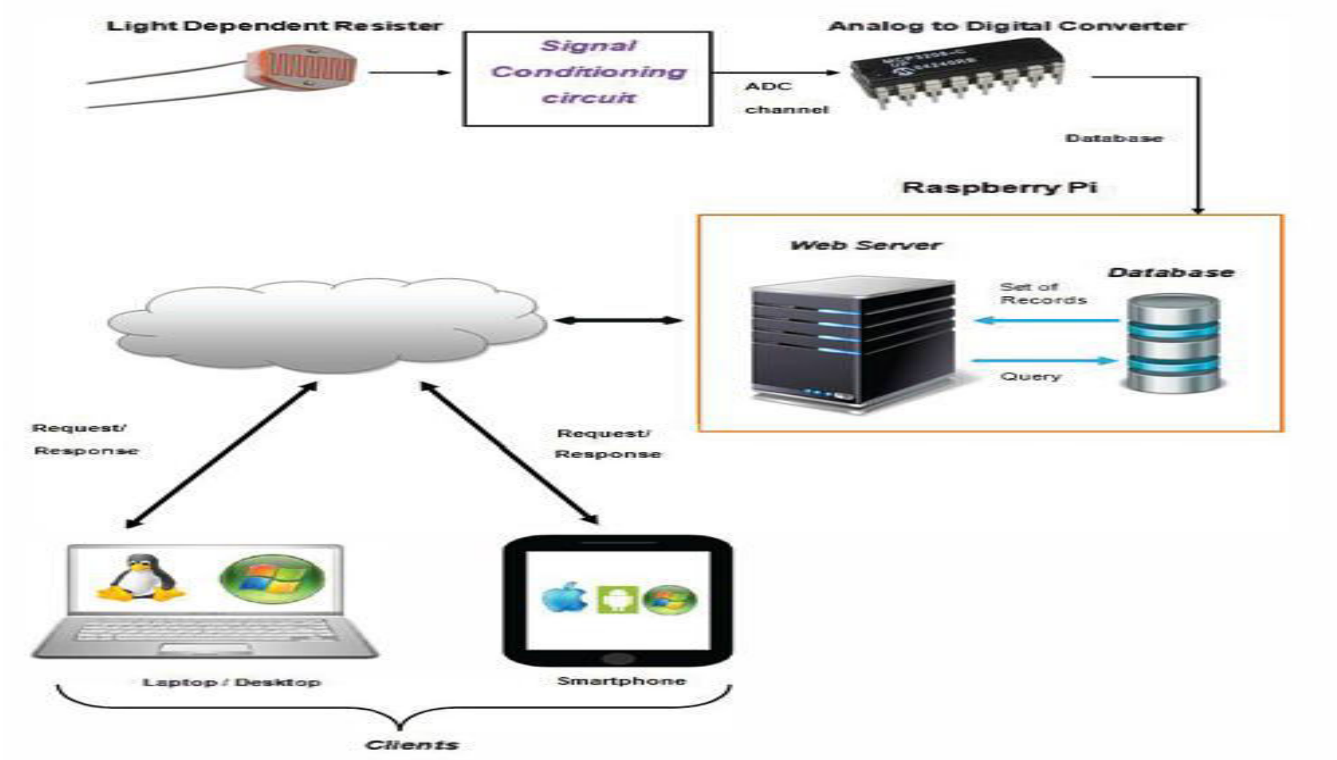


Fig.7 Cloud Based proposed model

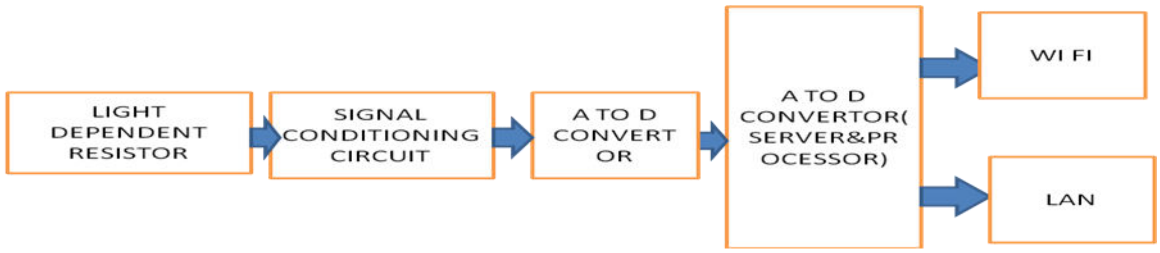


Fig.8 Block Diagram of proposed model

The hardware model of proposed system as shown in fig.8 includes also LED , in which the model consist of the output of LED Display through which it shows the message of the three conditions of the light intensity monitoring systems.

- Normal light intensity (40<ldr<900)
- Low light intensity (ldr<40)
- High light intensity(ldr>900)

V.RESULTS AND EVALUATION

experimental setup of the proposed model is shown below, which consist of display section, indicating section and monitoring section as shown in fig.9

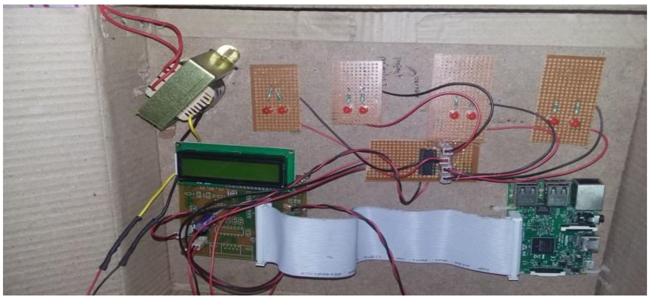


Fig.9 Experimental setup

In this figure10 section the low intensity light has been monitored which show the reading in the display and indicated by the LED lights all had glowed.

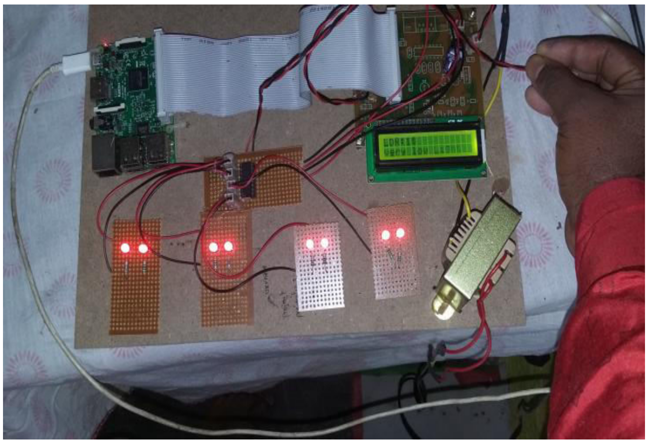


Fig.10 Experiment work for low light intensity (LDR<40)

In this figure11 section the normal intensity light has been monitored which show the reading in the display and indicated by the LED lights, only one LED had glow

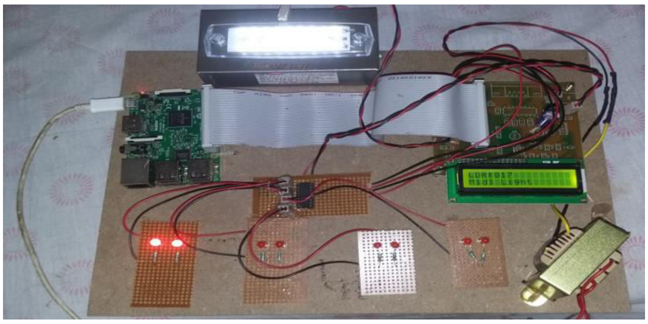


Fig.11 Experiment work for Normal light intensity (40<LDR<900)

In this figure12 section the high intensity light has been monitored which show the reading in the display and indicated by the LED lights

A. RESULTS

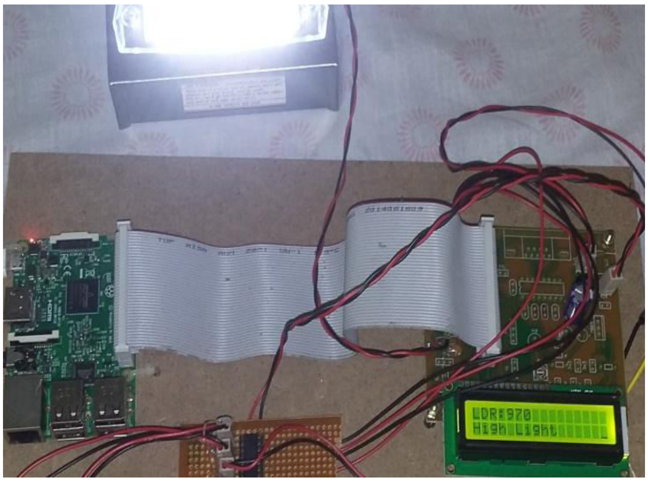


Fig.12 Experiment work for high light intensity (LDR<900)

EVALUATION

These conditions can be made by using the proposed model in such a way that it can also be used as cloud based monitoring system which can update the third party server about the conditions using the proposed system which is shown below

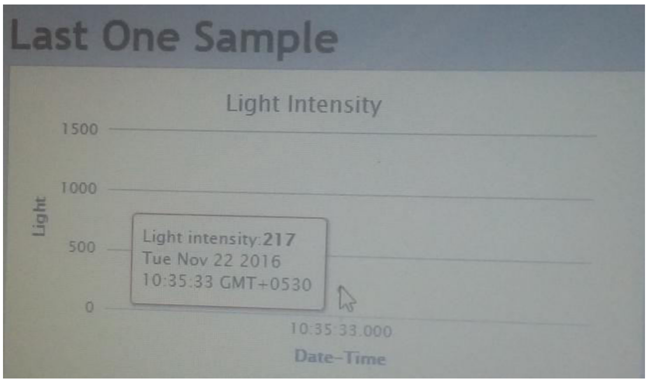


Fig.13 Graphical data monitoring and send to 3rd party server

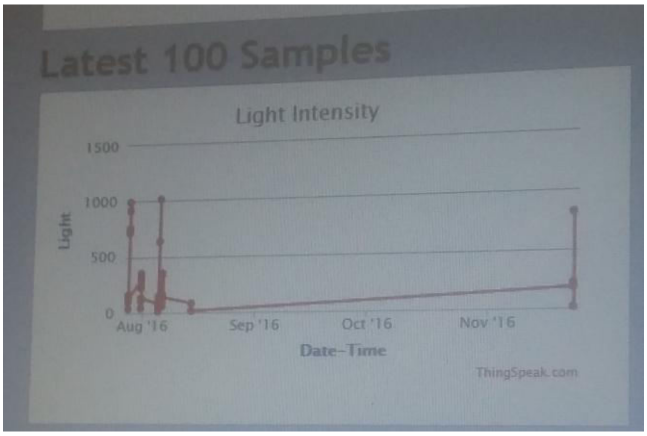


Fig.14 Graphical data of light intensity monitoring of four months

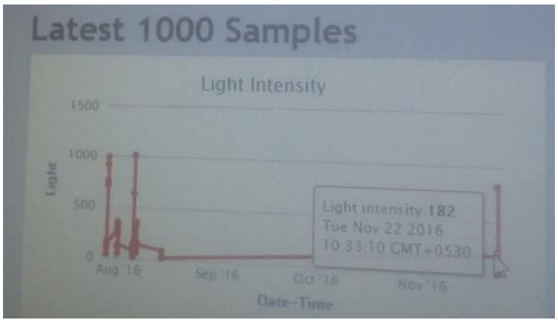
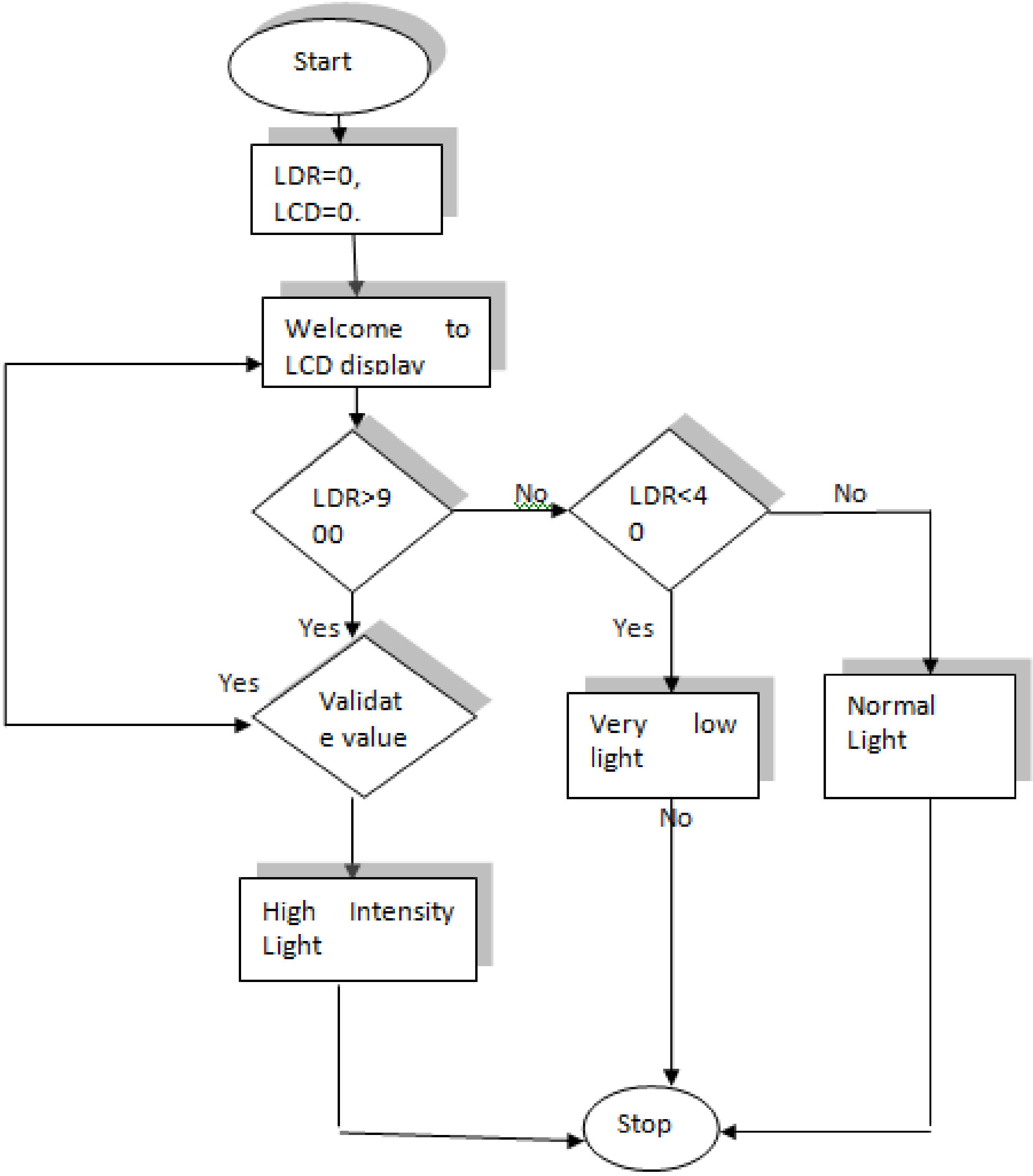


Fig.15 Graphical data of light intensity monitoring of four months on Nov 22-2016

Graphical results shows the instantaneous light intensity graph. If we observe the Figs 13 &15 in between Aug, Sep, Oct& Nov, light intensity recorded the average light intensity (217 LUX & 182 LUX). It is also showing maximum light intensity, minimum light intensity with date, time. Fig.14 shows 4 months report , it helps data analyst to understand average light intensity of each day.

B. Flow chart of the proposed model



VII. CONCLUSION AND FUTURE WORK

The Facility manger will have skill, training and experience but lagging with lack of information to take action immediately. In the paper, we have proposed and developed cloud based light intensity monitoring system. This helps to Facility manger to take necessary action at right time, with proper controlling with can achieve desired results. To evaluate the system, we have considered laboratory as an example but it can be used at various applications like traffic light monitoring, poultry lighting and museum lighting etc. to avoid damage.

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STACK FORCING AND MTCMOS LEAKAGE POWER REDUCTION TECHNIQUES IN CARRY SELECT ADDER

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Abstract: **Low power has emerged as a principal theme in today's electronics industry. The need for low power has caused a major paradigm shift where power dissipation has become an important consideration as performance and area.In the past, the major concerns of the VLSI designer were area, performance, cost and reliability; power consideration was mostly of only secondary importance. In recent years, power is being given comparable weight to area and speed considerations. Several factors have contributed to this trend. Designing of power-efficient and high speed digital logic systems is very crucial and important task. In digital adders, the speed of addition is limited by the time required to transmit a carry through the adder. Carry Select Adder (CSLA) is one of the fastest adders used in many data-processing processors to perform fast arithmetic functions This work uses a simple and an efficient gate-level modification which drastically reduces the area of CSLA. Based on this modification 16-bit Carry Select Adder (CSLA) architectures have been developed and compared with the regular CSLA architecture developed in CMOS process technology. The proposed design has reduced area to a great extent when compared with the previous CSLA developed. This work estimates the performance of the proposed designs with the regular designs in terms of area and power and is implemented in CADENCE tool. The results analysis shows that the proposed CSLA structure developed technology is better than the regular CSLA developed in CMOS technology. In this paper leakage reduction techniques such as stack forcing and MTCMOS are being implemented on CMOS 16 Bit Carry Select Adder Digital Circuit**

***Designs and simulations are being done on*: CADENCE 90NM Technology**

I. INTRODUCTION

With the rapid progress in semiconductor technology, chip density and operation frequency have increased, making the power consumption in battery-operated portable devices a major concern. High power consumption reduces the battery service life. The goal of low-power design for battery-powered devices is thus to extend the battery service life while meeting performance requirements. Reducing power dissipation is a design goal even for non-portable devices since excessive power dissipation results in increased packaging and cooling costs as well as potential reliability problems. IC power dissipation consists of different components depending on the circuit operating mode. First, the switching or dynamic power component dominates during the active mode of operation. Second, there are two primary leakage sources, the active component and the standby leakage component. The standby leakage may be made significantly smaller than the active leakage by changing the body bias conditions or by power-gating.

II. LEAKAGE POWER REDUCTION TECHNIQUES:

A. STACK FORCING TECHNIQUE:

Leakage currents in NMOS or PMOS transistors depend exponentially on the voltage at the four terminals of transistor. Increasing the source voltage, VS of NMOS transistor reduces sub-threshold leakage current exponentially because of the following three effects:

1. Gate-to-source voltage becomes negative, thus the sub-threshold current reduces exponentially.
2. Due to negative body to source potential, body effect increases which results in increased threshold voltage and thus reducing the sub-threshold leakage.
3. As drain-to-source potential decreases, drain induced barrier lowering (DIBL) effect lowers and thus sub-threshold leakage decreases.

This phenomenon is also called self-reverse biasing of transistor. The self-reverse bias effect occurs when stack of transistors are turned OFF. The result is reduced leakage power consumption. Turning OFF more than one transistor in a stack increases the internal voltage (source voltage) of the stack, which acts as reverse biasing the source. By using input vector control approach, the leakage power can be reduced in naturally stacked transistor circuit such as CMOS NAND or NOR by providing input vector with least leakage power.

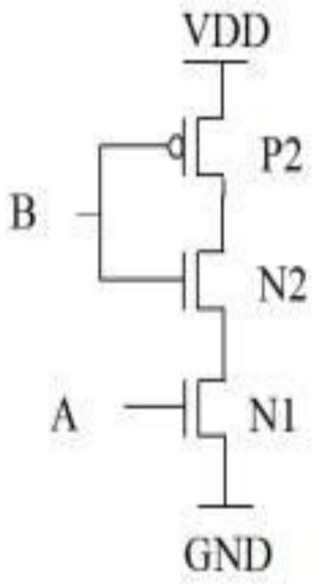

Fig 1: Stacked CMOS Inverter

Threshold voltage of a CMOS transistor can be controlled using body bias. In general, we apply Vdd to the body (e.g., an n-well or n-tub) of pMOS and apply Gnd to a body (e.g., p well or p-substrate) of nMOS. This condition, in which source voltage and body voltage of a transistor are the same, is called Zero-Body Bias (ZBB). Threshold voltage at ZBB is called ZBB threshold voltage. When body voltage is lower than source voltage by biasing negative voltage to body, this condition is called Reverse-Body Bias (RBB). Alternatively, when body voltage is higher than source voltage by biasing positive voltage to body, this condition is called

Forward-Body Bias (FBB). When RBB is applied to a transistor, threshold voltage increases, and when FBB applied to a transistor, threshold voltage decreases. This phenomenon is called body-bias effect, and this is frequently used to control threshold voltage dynamically.

The forced stack approach has been employed on various digital circuits. The Stack forcing is applied on the circuits where single transistor is present between the power supply and the output node of that stage. It has been observed that Stack forcing has been applied by replacing each of the transistors by two equal sized transistors; the two stack condition for a given Vt with the least delay is for Wu = Wl = ½ W, to reduce the active mode leakage power. Leakage power reduces but at the cost of large delay overhead. Hence, in this work, the forced stack has been employed selectively keeping in mind that the circuit performance does not degrade too much.

Disadvantages of Stack Forcing technique:

1. Affect Delay, area
2. Can only reduce leakage power in standby mode
3. Not suitable for sequential circuit.

B. MULTI THRESHOLD CMOS TECHNIQUE (MTCMOS)

In MTCMOS technique, a high-threshold voltage transistor is inserted in series with the power supply and the existing design and ground. In fact, only one type (either PMOS or NMOS) of high Vt transistor is sufficient for leakage reduction as shown in Fig.1 When In active mode, sleep transistors are turned ON, while in standby mode sleep transistors are turned OFF by applying appropriate OSFET increase area and delay. These high threshold voltage levels in the power gated mode, and large inserted M sleep transistors act as a current gate to the low threshold designed circuit, so this technique is also referred as Power Gating. Inserting additional sleep transistor(s) has an adverse effect on the circuit delay. Therefore, sizing sleep transistors is an important design consideration

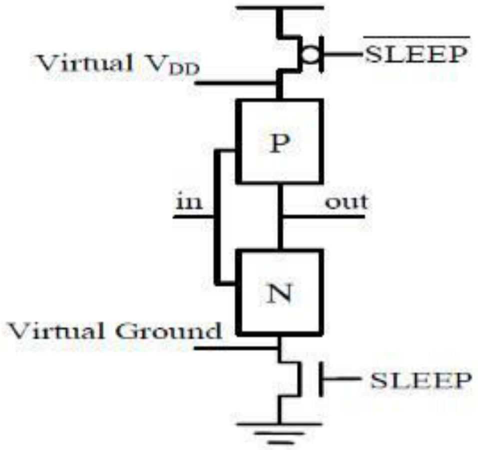

Fig 2: Power gating circuit

In the ACTIVE state, the sleep transistor is on. Therefore, the circuit functions as usual. In the STANDBY state, the transistor is turned off, which disconnects the gate from the ground. Note that to lower the leakage, the threshold voltage of the sleep transistor must be large. Otherwise, the sleep transistor will have a high leakage current, which will make the power gating less effective. Additional savings may be achieved if the width of the sleep transistor is smaller than the combined width of the transistors in the pull-down network. In practice,

Dual VT CMOS or Multi-Threshold CMOS (MTCMOS) is used for power gating. In these technologies there are several types of transistors with different VT values. Transistors with a low VT are used to implement the logic, while high-VT devices are used as sleep transistors. To guarantee the proper functionality of the circuit, the sleep transistor has to be carefully sized to decrease its voltage drop while it is on. The voltage drop on the sleep transistor decreases the effective supply voltage of the logic gate. Also, it increases the threshold of the pull-down transistors due to the body effect. This increases the high-to-low transition delay of the circuit.

This problem can be solved by using a large sleep transistor. On the other hand, using a large sleep transistor increases the area overhead and the dynamic power consumed for turning the transistor on and off. Note that because of this dynamic power consumption, it is not possible to save power for short idle periods. There is a minimum duration of the idle time below which power saving is impossible. Increasing the size of the sleep transistors increases this minimum duration.

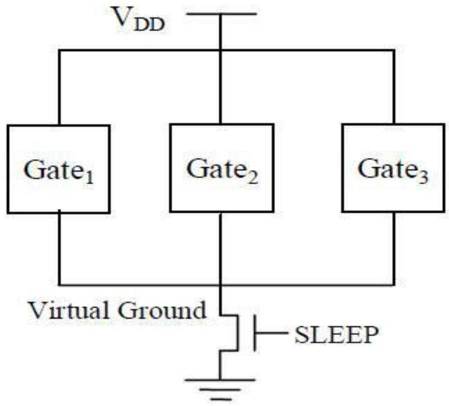

Fig 3:Using one Sleep Transistor for Several Gates

Since using one transistor for each logic gate results in a large area and power overhead, one transistor may be used for each group of gates. To find the optimum size of the sleep transistor, it is necessary to find the vector that causes the worst case delay in the circuit. This requires simulating the circuit under all possible input values, a task that is not possible for large circuits.

III. BASICS OF 16 BIT CARRY SELECT ADDER

Today's there are a growing number of portable applications requiring small-area low-power high throughput circuitry. Therefore, circuits with low power utilization grow to be the most important candidates for design of microprocessors and system mechanism. The battery technology does not advance at the same rate as the microelectronics technology and there is a imperfect quantity of power available for the mobile systems. The goal of extending the battery life span of portable electronics is to reduce the energy consumed per arithmetic operation, but low power consumption does not essentially imply low energy. To execute an arithmetic operation, a circuit can obtain through very low power by clocking at very low frequency but it may take a very long time to complete the operation. An adder is one of the most critical components of a processor which determines its throughput, and for address generation in case of cache or memory access. The full adder performance would affect the system as a whole.

In this paper, we propose a systematic approach to design 16-Bit Carry Select Adder using full adders and multiplexers.The adders are useful in larger circuits such as multipliers despite the threshold-loss problem. Power minimization is one of the primary concerns in today's VLSI design methodologies because of two reasons one is the long battery operating life requirement of portable devices and second is due to increasing number of transistors on a single chip leads to high power dissipation. In VLSI applications, 1-bit full adder cell is the fundamental gate used in many arithmetic circuits like adders and multipliers. Thus, increasing the performance of the full adder block leads to the enhancement of the overall system performance. A full adder has three inputs and two outputs block in which the outputs are the addition of three inputs. Basic fundamental units used in various circuits such as parity checkers, compressors and comparators are full adders. This technique uses high threshold voltage sleep transistor which cut-off a circuit block when the block is not switching.

In this paper I have implemented a 16bit Carry Select Adder (CSLA), in electronics; a CSLA is a particular way to implement an adder, which is a logic element that computes the n-bit sum of two n-bit numbers. CSLA is simple but rather a fast adder. CSLA generally consists of two ripple carry adders and a multiplexer. Adding two n- bit numbers with a CSLA is done with two adders (therefore two ripple carry adders) in order to perform the calculation twice, one time with the assumption of the carry being zero and the other assuming one. After the two results are calculated, the correct sum, as well as the correct carry, is then selected with the multiplexer once the correct carry is known.

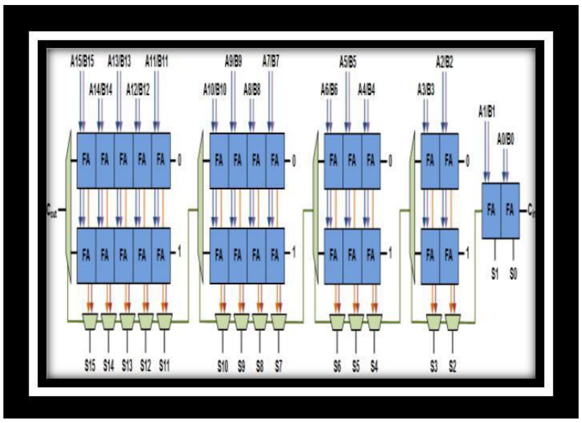

Fig 4: 16 Bit Carry Select Adder

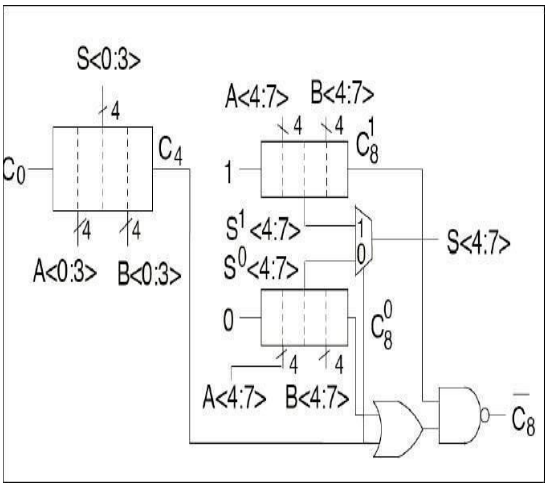

FIG 5: Internal Working Of CSLA

IV. IMPLEMENTATION RESULTS.

CMOS FULL ADDER

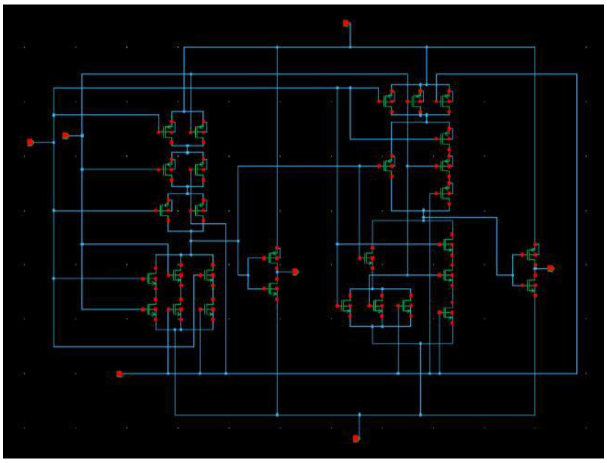

Fig 6: CMOS Full Adder Schematic

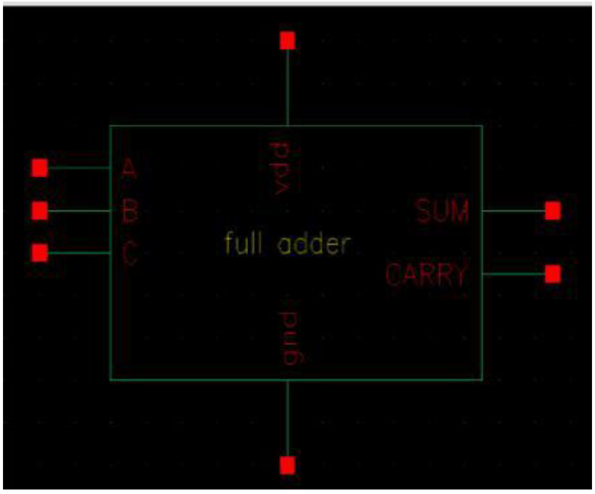

Fig 7: CMOS Full Adder Instance

MULTIPLEXER

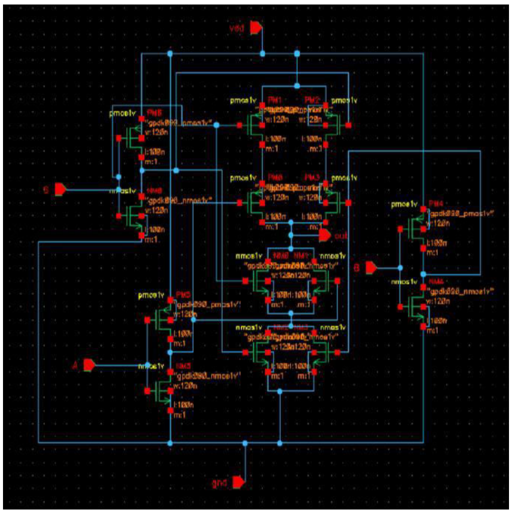

Fig 8:CMOS 2:1 Multiplexer Schematic

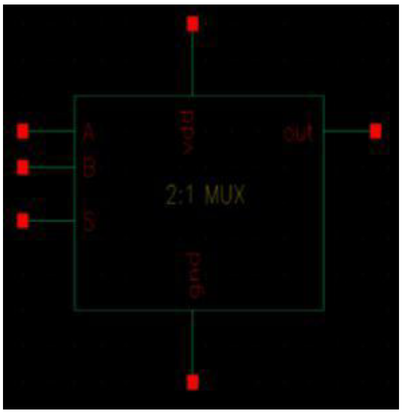

Fig 9: CMOS 2:1 Multiplexer Schematic

A. CARRY SELECT ADDER

SUM 0 TO SUM 15

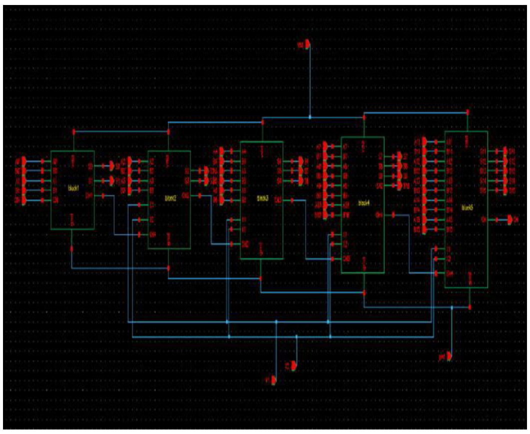

Fig 10: CMOS 16 bit carry select adder(sum0 to sum15) schematic

16 BIT CARRY SELECT ADDER TEST CIRCUIT

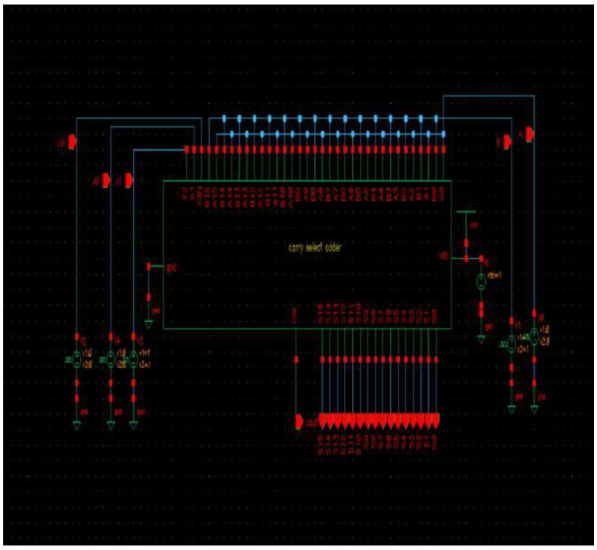

Fig 11: CMOS 16 bit carry select adder(sum0 to sum15) test circuit

B. 16 BIT CARRY SELECT ADDER POWER

16 BIT CSLA BASIC CIRCUIT (CMOS) POWER

Fig 12: CMOS 16 bit carry select adder(sum0 to sum15) test circuit power

16 BIT CSLA USING STACK FORCING TECHNIQUE POWER

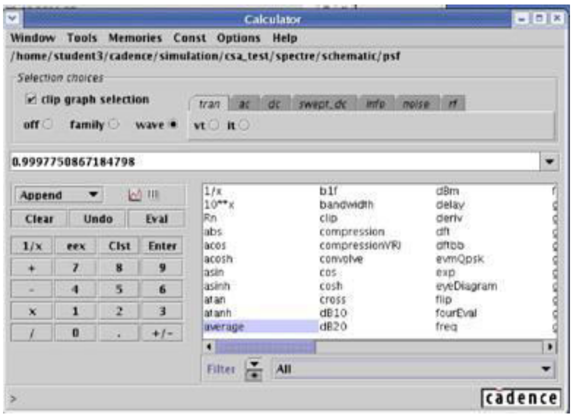

Fig 13: CMOS 16 bit carry select adder(sum0 to sum15) using stack forcing technique power

16 BIT CSLA USING MTCMOS POWER

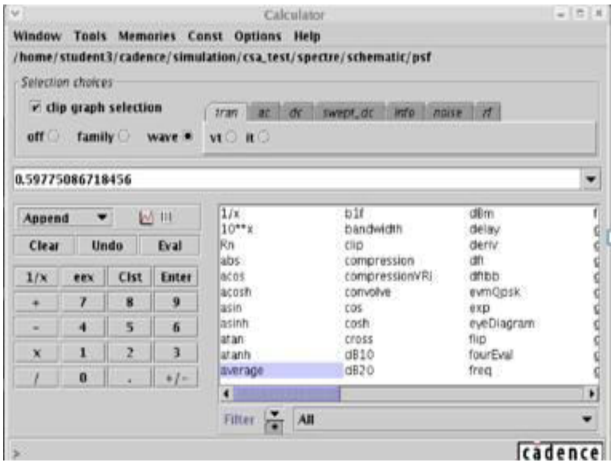

Fig 14: CMOS 16 bit carry select adder(sum0 to sum15) using MTCMOS technique power

TABLE1: 16 BIT CARRY SELECT ADDER POWER COMPARED IN DIFFERENT TECHNIQUES TECHNIQUE USED

TECHNIQUE USED	TOTAL POWER
BASIC CIRCUIT (CMOS)	3.80µW
STACK FORCING	0.99µW
MTCMOS TECHNIQUE	0.59µW

V. CONCLUSION

In this paper, a carry select adder has been designed using low power full adders and low power multiplexers. Those low power full adders have been designed. In this project power is reduced at transistor level.

As Low power has emerged as a principal theme in today's electronics industry. The need for low power has caused a major paradigm shift where power dissipation has become an important consideration as performance and area. In the past, the major concerns of the VLSI designer were area, performance, cost and reliability; power consideration was mostly of only secondary importance. In recent years, power is being given comparable weight to area and speed considerations. Several factors have contributed to this trend. so it is essential to reduce power.

Using CADENCE tool the proposed circuit is made to run and simulated. The Power consumptions of both the Conventional and the proposed carry select adders are compared and the reduction in power consumption in the case of proposed carry select adder(CSLA) has been observed. The power consumption values have been tabulated.

The future scope of the project is one can reduce dynamic power by using other leakage reduction techniques at the transistor level in various other circuits. The entire design has been done in CADENCE TOOL 90nm Technology

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BLIND CHANNEL (BC) EQUALIZATION BASED ON FRACTIONAL SPACED CMA AND CS-LMS USING ADAPTIVE MMSE EQUALIZER

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Abstract: The adaptive (Self Adjusted) algorithm has been broadly used in the fields of Stochastic and Advanced digital signal processing like channel estimation, channel equalization, noise cancellation, and so on. One of the most significant adaptive algorithms is the NLMS algorithm. In this paper several objective optimization approaches to fast blind channel equalization are existing with different filter structures are presented to nullify the diverse form of noises. A Novel idea in this paper is the constrained stability least mean square (CSLMS) algorithm is proposed which has the distinct feature of minimizing mean square error (MMSE) to effectively. Different adaptive filter methods are proposed and finally the CSLMS based adaptive filter algorithm is found to be the best in performance in terms of signal to noise ratio and mean square error without increasing the computational complexity.

Keywords: Normalized Least mean square (NLMS), Constant modulus algorithm (CMA), constrained stability least mean square (CSLMS).

I. INTRODUCTION

There are many digital signal processing applications in which second order statistics cannot be specified. Such application includes channel equalization echo cancellation and noise cancellation. In these applications, filters with adjustable coefficients called Adaptive filters are employed. An adaptive filter is a filter that self adjusts its transfer function according to an optimizing algorithm. It adapts the performance based on the input signal. Such filters incorporate algorithms that allow the filter coefficients to adapt to the signal statics. There are different approaches used in adaptive filtering, which are as follows:

Adaptive techniques use algorithms, which enable the adaptive filter to adjust its parameters to produce an output that matches the output of an unknown system. This algorithm employs an individual convergence factor that is updated for each adaptive filter coefficient at each iteration.

Blind equalization has the potential to improve the efficiency of communication systems by eliminating training signals. Difficulties of its application in wireless communications, however, are due largely to the characteristics of the propagation media multipath delays and fast fading. The challenge is achieving blind equalization using only a limited amount of data.

The key idea of this paper is to combine the approach based on minimizing the constant modulus cost and that based on matching the second-order cyclostationary statistics. The main feature of the proposed approach **is** the improved convergence property over the standard CMA equalization and the improved robustness for ill-conditioned channels.

II. BLIND CHANNEL (BC) EQUALIZATION

The field of blind channel equalization has been in existence for over a decade now. Research has focused on developing new algorithms and formulating a theoretical justification for these algorithms. Blind channel equalization is also called as a self-recovering equalization. The aim of blind equalization is to recover the unknown input sequence to the unknown channel based solely on the probabilistic and statistical properties of the input sequence. The receiver can coordinate to the received signal and to adjust the equalizer without the reference sequence. The term blind is used in this equalizer because it performs the equalization on the data without a reference signal. Instead, the blind equalizer depends on knowledge of the signal structure and its statistic to perform the equalization. A natural question for direct adaptive equalization with reference is, ``How can we adapt our filter F, without the use of a reference signal''. There has been extensive research on this subject for single user applications as well as multi-user applications. The Constant Modulus Algorithm is an algorithm employed for the blind adaptation problem. Fractional spaced CMA used to directly estimate equalizer *f.* It is similar to CMA. In fractional space it is global convergences.

$$\min J = E\left[(\left|f^H X(n)\right|^2 - R_Z)^2 \right] \qquad (1)$$

Update rule:

$$f_{n+1} = f_n - \mu E(\left|f^H X(n)\right|^2 - R_Z) X(n) X^H(n) f_n \quad (2)$$

Algorithm:

1. Construct the received the sample
2. Construct the sample vector(n);
3. For n=1, 2…..
update function
4. Check the SER

III. CONSTANT MODULUS ALGORITHM (CMA)

The CMA, proposed by Godard, is the most popular technique for blind equalization. Consider the base band model of a digital communication channel characterized by finite impulse response (FIR) filter and additive white Gaussian noise. In order to remove effect of channel distortion, we use the equalizer to eliminate this effect. The cost function of CMA is defined as:

$$D_k^{(2)} = E\left[\left(\left|x_k\right|^2 - R_2 \right)^2 \right] \qquad (3)$$

In this equation R2 is reference signal depending on statistical properties of input signal x_k. It is defined as:

$$R_2 = \frac{E\left[\left|a_k\right|^4\right]}{E\left[\left|a_k\right|^2\right]} \qquad (4)$$

Using a stochastic gradient algorithm like LMS, we obtain updated equation of CMA:

$$C(k+1) = C(k) - \lambda X^*(k)\, y_k \left(\left|y_k\right|^2 - R_2 \right)$$

Where λ is the step-size parameter and the asterisk denotes complex conjugation. Error signal of CMA is**:**

$$e_k = y_k \left(\left|y_k\right|^2 - R_2 \right) \qquad (6)$$

For CMA update equation (5), assuming $e_k = 0$ at perfect equalization, we have $\left|y_k\right|^2 - R_2 = 0$ .It means CMA attempts to make the equalizer output lie on the circle with radius $\sqrt{R_2}$. Since the cost function is based only on the equalizer output modulus, so CMA is the phase blind algorithm. Furthermore, if the frequency offset exists in equalizer output, the output constellation will spin. Godard demonstrated that the cost function can be applied even to non-constant modulus signals such as rectangular QAM constellations.

IV. ADAPTIVE MMSE

The signal after MMSE can be expressed in matrix form as

$$\hat{s}(i) = w^H y(i) \qquad (7)$$

Where $y(i) = H^T(i) s(i) + n(i)$

M is the length of the MMSE Equalizer; Equalizer coefficient vector is given by

$$w = \left[w_1, w_2, w_3, \ldots. w_M \right]^T \qquad (8)$$

The error signal e (i) is given as

$$e(n) = d(i) - \hat{s}(i) \qquad (9)$$

Where d (i) is the desired response for MMSE Equalizer

d(i) =s (i+D)

D is the delay parameter which is usually L+1.

The MMSE criterion is used to derive the optimal equalizer coefficients vector *w:*

$$w = \min imizeE\left\{ \left|e\right|^2 \right\} \qquad (10$$

We make assumption that signal s(i) and noise n(i) are independent identify distribution stochastic variable and uncorrelated each other, then the equalizer coefficient vector w can be expressed as

$$w = (H^H H + \frac{1}{SNR} I)^{-1} H^H \delta_D \qquad (11)$$

Where

$$\delta_D = [0.......1_D;0........0]^T_{1X(L+M-1)} \qquad SNR = \frac{\sigma_s^2}{\sigma_n^2}$$

denotes the signal noise ratio I is MxM identity matrix.

To reduce the complexity caused by matrix inversion of ideal MMSE equalizer, we propose an adaptive MMSE equalizer algorithm. In code-multiplexed pilot CDMA systems, conventional adaptive equalizer is difficult to implement for lack of reference signal. The steepest descent method is used to derive adaptive equalizer algorithm in code-multiplexed pilot CDMA systems.

Hence the mean square error J can be expressed as

$$J(w) = E\left[e(i)e(i)^*\right] = \sigma_s^2 - w^H p - p^H w + w^H R w$$

(12)

Where autocorrelation matrix $R = E[y(i)y^H(i)]$ correlation vector $p = E[y(i)d^*(i)]$, σ_s^2 denotes the signal power; (.)* represents conjugate operation. Because the wireless channel is time-varying, the equalizer coefficients vector w must be updated real time. Conventional adaptive algorithm requires reference signal d (i), while in the downlink of code-multiplexed pilot CDMA systems, d (i) is difficult to distill. To resolve this problem, the steepest decent method is used. From Eqn.8, the gradient vector is $\frac{\partial J(w)}{w} = -2p + 2Rw$ then the equalizer coefficients updating equation is

$$w(i+1) = w(i) + 2\mu[p - Rw(i)] \qquad (13)$$

Where parameter μ is a positive real-valued constant which controls the size of the incremental correction applied to the equalizer coefficients vector.

For the autocorrelation matrix:

$$R = E[y(i)y^H(i)]$$

$$R = E[s(i)s^H(i)]\left\{H^H(i)H(i)\right\}^T + E[n(i)n^H(i)]$$

$$R = \sigma_s^2\left\{H^H(i)H(i)\right\}^T + \sigma_n^2 I \qquad (14)$$

The cross-correlation vector

$$p = E[y(i)d^*(i)] = E[(H^T(i)s(i) + n(s))s^*(i-D)]$$

$$p = \sigma_s^2 H^T(i)\delta_D$$

(15)

From Eqn.9, 10, 11, we can obtain the time recursive equation of MMSE equalizer by:

$$w(i+1) = w(i) + 2\mu\sigma_s^2[H^T(i)\delta_D - (\left\{H^H(i)H(i)\right\}^T + \frac{1}{SNR} I w(i)]$$

(16)

As can be seen from Eqn.15, the updating process avoids the matrix inversion operation. On the other hand, the updating process withholds the requirement to store the autocorrelation matrix **R** (*i*) and only the equalizer coefficients vector of last time is needed. From Eqn.15 we know, the channel convolution matrix **H**(*i*) is required to update the equalizer coefficients vector.

For CMA , channel response can be estimated through code- multiplexed pilot. In this paper, the low complexity sliding- window method is used to estimate the channel coefficients, which can be expressed as

$$\hat{\beta}_l(i) = \frac{1}{2\sqrt{\alpha pw(i+1)T_s}} \int_{\tau_l(i+\frac{w}{2})T_s}^{\tau_l(i+\frac{w}{2})T_s} y(t)c_p^*(t-\tau_l)dt$$

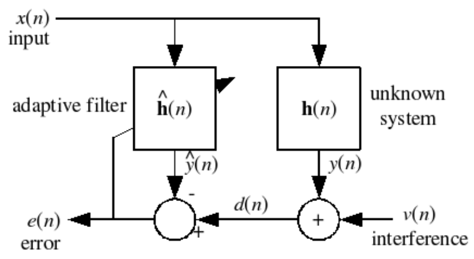
(17)

Where β_l (i) is estimation of the complex gain of l^{th} path; w is the length of sliding-window in symbols and should be selected properly according to the varying speed of the channel.

V.NLMS Algorithm

Usually, the adaptive algorithm consists of a transfer filter for processing the input signal and an algorithm unit for update the transfer filter's coefficients, x (n) the input signal;$w(n) = [w_0, w_1, w_2, ...w_l]$ is the vector of the transfer filter coefficient; d(n) is the desired function of the transfer filter; y(n) is the output of transfer function $e(n)$is the error value, and it can be written as:

$$e(n) = d(n) - y(n) \qquad (18)$$



The Adaptive algorithm unit represents some algorithm to update the coefficients of the transfer filter. For LMS algorithm, the method to update the coefficients of the transfer filter is given as follows:

$$w(n) = w(n+1) + \mu^{*} x(n) * e(n) \qquad (19)$$

μ , is the step of LMS Algorithm.

The main drawback of the "pure" LMS algorithm is that it is sensitive to the scaling of its input $x(n)$. This makes it very hard (if not impossible) to choose a learning rate μ that guarantees stability of the algorithm. The *Normalized least mean squares filter* (NLMS) is a variant of the LMS algorithm that solves this problem by normalizing with the power of the input. The NLMS algorithm can be summarized as:

Parameters: p = filter order μ = step size

Initialization

$$\hat{h}(0) = 0$$

Computation: For n = 0,1,2,...

$$X(\mathrm{n}) = [x(n), x(n-1), \ldots . x(n-p+1)]^{T} \qquad (20)$$

$$e(n) = d(n) - h^{\wedge H}(\mathrm{n})\,\mathrm{X(n)} \qquad (21)$$

$$\hat{\mathrm{h}}(\mathrm{n}+1) = \hat{h}(\mathrm{n}) + \frac{\mu e^{*}(n) X(\mathrm{n})}{X^{H}(\mathrm{n})\,\mathrm{X(n)}}$$

(22)

The major drawback of the LMS and NLMS algorithms is the large value of mean square error which results in the distortion in the enhanced ECG signal. In the CSLMS algorithm the step size parameter is inversely proportional to the squared norm of the difference between the two consecutive inputs rather than the input vector as in NLMS algorithm.

The weight update relation of the CSLMS is as follows

$$w(n+1) = w(n) + \mu \left[\frac{\delta_x(n)\delta_e(n)}{\|\delta_x(n)\|^2} \right] \qquad (23)$$

VI. Experimental and Analysis

The performances for the Adaptive MMSE and adaptive CMA algorithm through and CSLMS algorithm is experimental performed with accurate figures from 1-4.

The performances of the channel estimation is an analyze in such a way that transmitted bits and receiver bits, Equalizers and very important task that is Convergences.

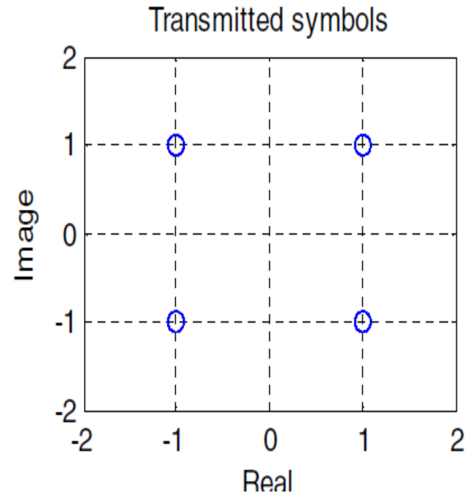


Fig.1.Transmitter symbols of Fractional-CMA

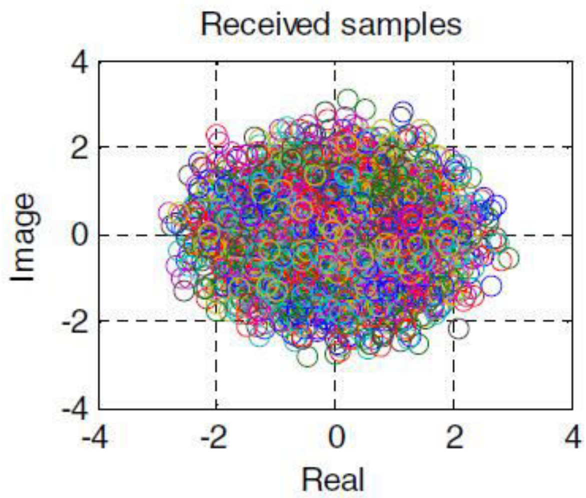


Fig.2: Receiver sample

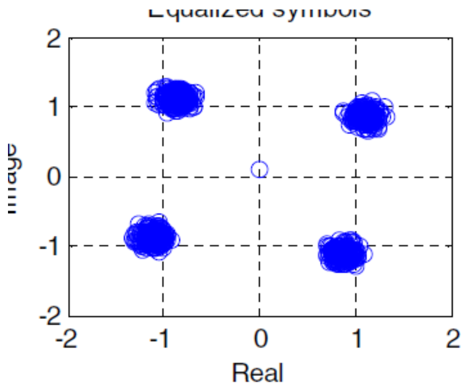


Fig..3. Adaptive Fractional- CMA Equalizer

The first figures from 1- 3 are obtained for Adaptive CMA Equalizer, with more efficient for equalization and convergences. Secondly from figure 4-5 are obtained for an Adaptive MMSE equalizer through CSLMS algorithm. The efficient of equalization and convergences is too good. The time complexity is very less and more efficient for advance communication systems.

Here the performance comparision of different adapative algorithms has been analysed.It concludes that the performance of RLS adapative algorithm is high as compared to other algorithms due to the less mean-square error(MSE) and high stability.

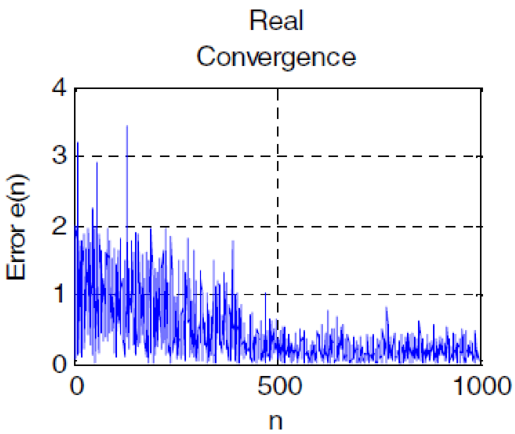


Fig.4: Adaptive MMSE Equalizer through CSLMS

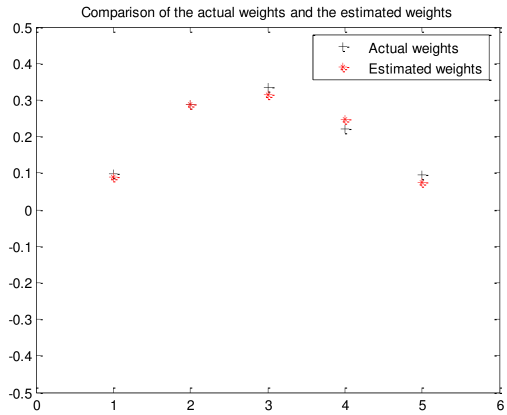


Fig..5: Comparison of the actual weight and estimated weight

VII. CONCLUSION

It concludes that CSLMS algorithm shows the improved performance in terms of SNR and at the same time reducing the mean square error and the misadjustment compared to the conventional methods such as LMS and normalized LMS algorithms. The simulation result shows that the performance of CSLMS algorithm is better than the NLMS algorithm.

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PERFORMANCE IMPROVEMENT ANALYSIS OF 4G SYSTEMS

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Abstract—**The great demand for wider range of services spanning from voice to high rate data services required for supporting 4G mobile communications. This leads to transformation of the 3G to 4G reliable communication in terms of data rate and spectral efficiency. The design issues and the effects of the multipath induced channel quality fluctuations on the performance will be addressed and the investigation is performed to improve the performance through proposed link adaptation EMDDCE for the OSTBC MIMO-OFDM. At bit error rate (*BER)* 0.1, the performance advantage of approximately 4.8*dB* is observed over the initial estimation algorithm in terms of Eb/No. The performance of proposed technique approaches Cramer Rao Lower Bound (CRLB) at about Eb/No is 14 *dB*. Therefore at high value of Eb/No, performs better than data aided (DA) algorithm when we consider CRLB.**

Keywords—BER; CRLB; Multi Input Multi Output; Orthogonal Frequency Divison Multiplexing.

I. INTRODUCTION

Multicarrier OFDM based wireless technique, used to combat frequency selective fading due to its robustness especially for high-speed data transmission. Combining OFDM with MIMO yields synergistic effects, such as enhanced reliable communication against frequency-selective fading and high scalability in possible data-transmission rates. To further improve the performance of 4G systems using MIMO OFDM, link adaptation of each sub channel, has extensively been investigated[1-4]. Link adaptation exploits the frequency-selectivity nature of wideband channels in a sub channel by sub channel basis and may sometimes result in a signaling overhead. A historical review on the link adaptation has been addressed in [5-8] and the references therein.

Channel state information (CSI) is required for link adaptation at the transmitter, which is usually estimated at the receiver and fed back to the transmitter. The overhead of CSI feedback is sometimes huge for MIMO-OFDM systems. Therefore, CSI feedback reduction for MIMO-OFDM systems has become an important research topic in the past several years. Based on pre-coding approaches for MIMO systems [9-12] have been obtained for MIMO-OFDM systems.

In literature, there are training-based, blind and semi-blind methods are referred as channel estimation techniques. The training-based method requires extra bandwidth to accommodate the periodic known symbols and thus reduces the spectral efficiency [13]. The blind method saves the spectral efficiency by utilizing the statistics of received signals. But, this method requires a large amount of received signals to obtain accurate statistics [14]. Semi- blind methods, on the other hand, combine the blind method with few pilot symbols to solve the ambiguity problem occurred in blind methods [15-16]. The accuracy of channel estimation directly affects the performance of MIMO-OFDM systems. Therefore, channel estimation is an important topic to facilitate MIMO-OFDM for wireless communications.

Link adaptation is a process to select the best transmission parameters to be employed for the next transmission, a reliable estimation of the channel state information (CSI) is necessary based on the prediction of the channel conditions. The proposed joint channel estimation algorithm Expectation Maximization Decision Directed Channel Estimation algorithm (EMDDCE) employing low complexity single user detection facilitates better performance advantage over the initial estimation algorithm in terms of Eb/No.

It can be inferred that the proposed EMDDCE based estimation method alleviates the number of computations and number of iterations in comparison to the traditional training based method. The bit error rate performance advantages of EMDDCE is are marginally better than existing algorithms.

II. SYSTEM MODEL

From figure, Space-Time Coding (STC) scheme is used to map the source symbols to the transmit antennas. The IFFT and the FFT are used for, modulating and demodulating the data constellations on the orthogonal Sub Carriers SCs.

At the input of the IFFT, number of data constellation points can be taken according to QAM signaling set (symbol mapping). The N output samples of the IFFT, being in time domain form the baseband signal carrying the data symbols on a set of N orthogonal SCs.

Due to the cyclic prefix, the transmitted signal becomes periodic, and the effect of the time-dispersive multipath channel becomes equivalent to a cyclic convolution, discarding the CP at the receiver. SCs remain orthogonal.

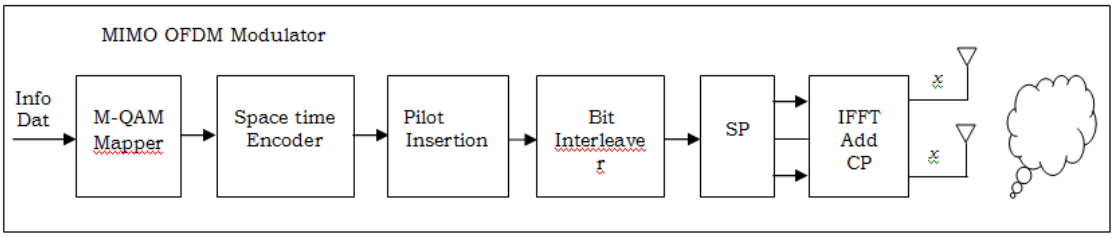

Fig.1.: Block diagram of MIMO-OFDM modulator

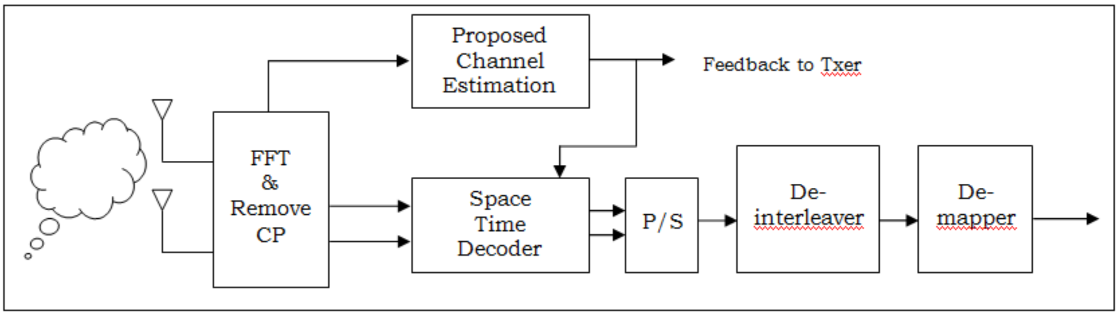

Fig.2.: Block diagram of MIMO-OFDM Demodulator

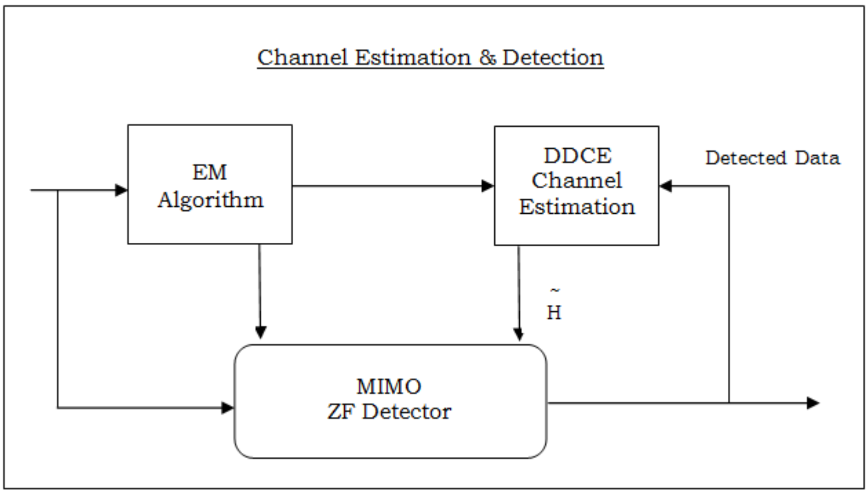

Fig.3.: Block diagram of proposed channel estimator

The wireless system can be represented as,

$$Y = HX + N \tag{1}$$

Where, H= fading channel coefficient

X= Transmitted symbol vector

Y= Received symbol vector

N= white Gaussian noise with mean zero and variance σ^2

Proposed EMDCE channel estimation method is presented below.

Step 1: Compute the expected value of the log-likelihood function of H by taking expectation over X, conditioned on Y as follows:

$$\ln\left(f(Y|H,X)\right) = \ln\left(\sum_{i=1}^{C}\frac{1}{C}\frac{1}{\sqrt{2\pi\sigma^2}}\exp\left(-\frac{1}{2\sigma^2}|Y-H\cdot X|^2\right)\right) \tag{2}$$

Then, by using the latest estimate of H denoted as $H^{(p)}$, as follows

$$Q(H \mid H^{(p)}) = E_x\{f(Y,X \mid H) \mid Y, H^{(p)}\} \tag{3}$$

Step 2: $H^{(p+1)}$ is determined by maximizing over all possible values of H as:

$$H^{(p+1)} = \arg\max_{H} Q(H \mid H^{(p)}) \tag{4}$$

The first iteration is conventional. The transmitter transmits a training packet/pilot/known data to the receiver. The receiver receives the training packet then estimates a channel quality using the training packet.

Step 3: The received signal is then equalized and channel decoded.

Step 4: The decoded decisions are then interleaved back to the channel estimator, which begins the next iteration.

Step 5: The channel estimator can now use the whole burst(both data and training bits) as known and re-estimate CIR.

Step 6: We use Least Mean Square (LMS) adaptation rule here to avoid heavy computations.

$$H(k+1) = H(k) + \Delta\varepsilon_k Y_k \tag{5}$$

Where

$$\varepsilon_k = d_k - \hat{d}_k$$

refers error between bit decisions and training sequence , Δ=step size.

III. SIMULATION PARAMETERS

The whole wireless channel transmission bandwidth is 800 kHz, and it is divided into 64 subcarriers (or tones). To make tones orthogonal to each other, symbol period is selected to be 80 microseconds. An additional 20 microseconds CP (N_{CP}=16) is utilized to give protection from ISI and ICI because of wireless channel delay spread. Therefore, total OFDM block length is *Ts* 100 microseconds and sub channel symbol rate is 10 kbaud.

The modulation scheme utilized in the proposed system is M-QAM. For those OFDM blocks containing pilot symbols, initial estimate of CSI is obtained from the wireless channel estimate of previous OFDM block. The fade rate is set 0.05 $f_d.Ts$ and for the following outcomes with maximum Doppler spread 500 f_d *Hz* .

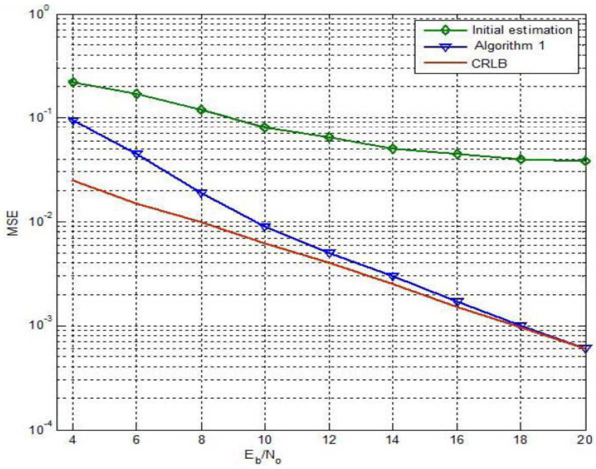

Fig.4: MSE versus SNR (dB) using DDCE

From figure 4, under time-varying environment (non stationary), Decision Directed Channel Estimation estimator (Algorithm1) outperforms the conventional initial estimation algorithm i.e., training based channel estimator method in terms of lower mean squared channel estimation error. This performance advantage gets elevated with the increasing value of *Eb/No*. At *MSE* 0.1, the performance advantage of approximately 4.3*dB* is observed by Algorithm1 over the initial estimation algorithm in terms of Eb/No. The performance of Algorithm1 approaches CRLB at about 20dB.

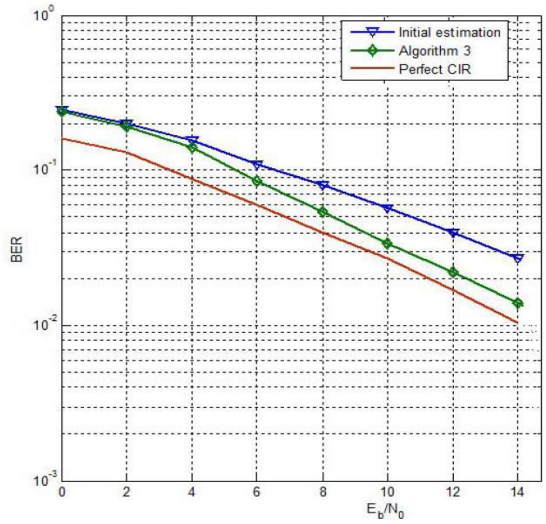

Fig.5: BER versus Eb/No (dB) using EM

From figure 5, under time-varying environment (non stationary), Algorithm 2 (EM) supersedes the initial estimation algorithm. This performance advantage gets elevated with the increasing value of *Eb*/*No*. At *BER* 0.03, the performance advantage of approximately 3.5*dB* is achieved by Algorithm2 over the initial estimation algorithm in terms of *Eb*/*No*.

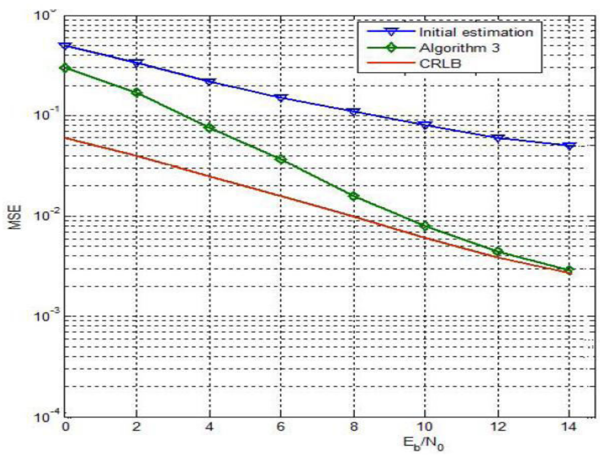

Fig.10: BER versus Eb/No (dB) using EMDDCE

Under time-varying environment (non stationary), proposed EMDDCE (Algorithm3) based wireless channel estimator outperforms the initial estimation algorithm in terms of lower mean squared channel estimation error. This performance advantage gets elevated with the increasing value of Eb/No.

At *MSE* 0.1, the performance advantage of approximately 4.8*dB* is observed by Algorithm3 over the initial estimation algorithm in terms of Eb/No. The performance of Algorithm3 approaches CRLB at about Eb/No is 14 *dB*. Therefore at high value of Eb/No, Algorithm3 performs better than Algorithm1 when we consider CRLB.

Through MATLAB simulation, it can be inferred that EMDDCE based estimation method alleviates the number of computations and number of iterations in comparison to the traditional EM method. The bit error rate performance advantages of EM and EMDDCE algorithms are marginally better than EM.

IV. Concluson And Future Scope

The simulation results reveal that the proposed algorithm EMDDCE, converges at a rate independently to the multipath delay spread. By passing the wireless channel estimate using demodulating transmitted symbols straightforwardly. Benefits and limitations of every algorithm have been explored by means of simulation. And, the proposed algorithm can accomplish near optimal estimates after a few iterations.

The performance can be further refined by exploiting the intelligent learning based channel coding schemes to reduce the BER on a burst by burst basis.

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A SURVEY ON BLIND SUPER RESOLUTION OF REAL-LIFE VIDEO SEQUENCES

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Abstract—**Super resolution (SR) for real-life video sequences is a challenging problem due to complex nature of the motion fields. In this paper, a novel blind SR method is proposed to improve the spatial resolution of video sequences, while the overall pointspread function of the imaging system, motion fields, and noise statistics are unknown. To estimate the blur(s), first, a nonuniform interpolation SR method is utilized to upsample the frames, and then, the blur(s) is(are) estimated through a multiscale process. The blur estimation process is initially performed on a few emphasized edges and gradually on more edges as the iterations continue. Also for faster convergence, the blur is estimated in the filter domain rather than the pixel domain. The high-resolution frames are estimated using a cost function that has the fidelity and regularization terms of type Huber–Markov random field to preserve edges and fine details. The fidelity term is adaptively weighted at each iteration usinga masking operation to suppress artifacts due to inaccurate motions. Very promising results are obtained for real-life videos containing detailed structures, complex motions, fast-moving objects, deformable regions, or severe brightness changes. The proposed method outperforms the state of the art in all performed experiments through both subjective and objective evaluations.**

Index Terms—Video super resolution, blur deconvolution, blind estimation, Huber Markov random field (HMRF).

I. INTRODUCTION

MULTI-IMAGE super resolution (SR) is the process of estimating a high resolution (HR) image by fusing a series of low-resolution (LR) images degraded by various artifacts such as aliasing, blurring, and noise. Video super resolution, by contrast, is the process of estimating a HR video from one or multiple LR videos in order to increase the spatial and/or temporal resolution(s). The spatial resolution of an imaging system depends on the spatial density of the detector (sensor) array and the point spread function (PSF) of the induced detector blur. The temporal resolution on the other hand, is influenced by the frame rate and exposure time of the camera . Spatial aliasing appears in images or video frames when the cut-off frequency of the detector is lower than that of the lens. Temporal aliasing arises in video sequences when the frame rate of the camera is not high enough to capture high frequencies caused by fast moving objects. The blur in the captured images and videos is the overall effect of different factors such as defocus, motion blur, optical blur, and detector's blur resulting from light integration within the active area of each detector in the array. The references provide overviews of different SR approaches.

One way to increase the resolution of a video is by overlaying a sliding window upon each frame and combining all frames falling inside the window to build the corresponding HR frame. Then the window slides to the location of the other frames and the process repeats. For this system to work, usually a local registration method (such as optical flow, block-based, pel-recursive, or Bayesian) is required to accurately estimate the displacement vector of each pixel or block within the frames. However, local registration may not be reliable in some cases, especially when there are complex dynamic changes (e.g. complex 3D motions), nonrigid deformations (e.g. flowing water, flickering fire), or changes in illumination .

Another class of single-video SR techniques is the one known as learning-based, patch-based or example-based video SR. The basic idea is that small space-time patches within a video are repeated many times inside the same video or other videos, at multiple spatio-temporal scales. Therefore, by replacing LR patches in the input video with equivalent HR patches from internal/external sources, the resolution can be improved. The major advantage of patch-based image/video SR methods is that motion estimation and object segmentation are not required. However, techniques of this group often have high computational complexity and most of them need offline database training. Furthermore, it is necessary that LR patches are generated from HR patches by a known PSF.

II. EXISTING METHOD

We present experimental results of digital super resolution (DSR) techniques on low resolution data collected using PANOPTES, a multi-aperture miniature folded imaging architecture. The flat form factor of PANOPTES architecture results in an optical system that is heavily blurred with space variant PSF which makes super resolution challenging. We also introduce a new DSR method called SRUM (Super-Resolution with Unsharpenning Mask) which can efficiently highlight edges by embedding an unsharpenning mask to the cost function. This has much better effect than just applying the mask after all iterations as a post-processing step.

DISADVANTAGE:

1. Not Applicable on motion blur videos
2. Less performance on high quality videos

III. PROPOSED METHOD

Super resolution (SR) for real-life video sequences is a challenging problem due to complex nature of the motion fields. In this paper, a novel blind SR method is proposed to improve the spatial resolution of video sequences, while the overall point spread function of the imaging system, motion fields, and noise statistics are unknown. To estimate the blur(s), first, a non-uniform interpolation SR method is utilized to upsample the frames, and then, the blur(s) is(are) estimated through a multiscale process. The blur estimation process is initially performed on a few emphasized edges and gradually on more edges as the iterations continue. Also for faster convergence, the blur is estimated in the filter domain rather than the pixel domain. The high-resolution frames are estimated using a cost function that has the fidelity and regularization terms of type Huber–Markov random field to preserve edges and fine details. The fidelity term is adaptively weighted at each iteration using a masking operation to suppress artifacts due to inaccurate motions. Very promising results are obtained for real-life videos containing detailed structures, complex motions, fast-moving objects, deformable regions, or severe brightness changes. The proposed method outperforms the state of the art in all performed experiments through both subjective and objective evaluations.

ADVANTAGES:

1. Applicable on motion blur videos
2. Better performance on high quality videos

DISADVANTAGE:

1. Robustness in sharpness is not much better.

In this paper the blind super resolution contains two parts

1.BLUR ESTIMATION.

2.FINAL FRAME ESTIMATION.

IV. BLUR ESTIMATION

In a multi-channel BD problem, the blurs could be estimated accurately along with the HR images. However in a blind SR problem with a possibly different blur for each frame, some ambiguity in the blur estimation is inevitable due to the downsampling operation. By contrast, in a blind SR problem in which all blurs are supposed to be identical or have gradual changes over time, such an ambiguity can be avoided. Moreover the assumption of identical (or gradually changing) blurs makes it possible to separate the registration and upsampling procedures from the deblurring process which significantly decreases the blur estimation complexity. In the NUI method to reconstruct the upsampled frame is explained. This upsampled yet-blurry frame is used to estimate the PSF(s) and the deblurred frames through an iterative alternative minimization (AM) process. The blur and frame estimation procedures are discussed . The estimated frames are used only for the deblurring process and so omitted thereafter. Finally, the overall AM optimization process is described.

Frame Upsampling:

We discuss the situations in which the warping and blurring operation are commutable.Although for videos with arbitrary local motions this commutability does not hold exactly for all pixels, however we assume here that this is approximately satisfied. The ultimate appropriateness of the approximation is validated by the eventual performance of the algorithm that is derived based on this model. With this assumption, (2) can be rewritten as:

gi =DMk,iHfk +ni =DMk,izk +ni **(1)**

where z_k = Hf_k is the upsampled but still blurry frame. Equation (4) suggests that we can first construct the upsampled frames z_k using an appropriate fusion method and then apply a deblurring method to z_k to estimate f_k and h. If noise characteristics are also the same for all frames, an appropriate way to estimate zk is using the NUI method. In NUI, the pixels of all LR frames are projected on to the HR image grid according to their motion fields, and then the intensities of the true locations on the grid are computed via interpolation. Our experiments show that using NUI for upsampling the frames leads to better estimates of f and h compared to when z_k is estimated iteratively from the LR frames g using a MAP (Maximum A Posteriori) or ML (Maximum Likelihood) method.

B. Frame Deblurring:

After upsampling the frames, we use the following cost function, J, to estimate the HR frames fk having an estimate of the blur h (or H):

$$\mathbf{J(f_{k)=||}\ \rho\ (Hf_k\text{-}z_k)||+\lambda^n\Sigma||\rho(\Delta_j f_k)||} \qquad \textbf{(2)}$$

where·1 denotes the l1 norm (defined for a sample vector x with elements xi as x= i |xi|), λn is the regularization coefficient, ρ(·) is the vector Huber function, ρ(·)1 is called the Huber norm, and ∇j (i = 1,...,4) are the gradient operators in 0°, 45°, 90° and 135° spatial directions. The first term in is called the fidelity term which is the Hubernorm of error between the observed and simulated LR frames. While in most works the l2-norm is used for the fidelity term, we use the robust Huber norm to better suppress the outliers resulting from inaccurate registration. The next two terms are the regularization terms which apply spatiotemporal smoothness to the HR video frames while preserving the edges.

Each element of the vector function ρ(·) is the Huber function defined as:

$$p(x)= \quad x^2 \qquad \text{if } |x|\leq T$$

$$2T|x|-T^2 \quad \text{if } |x|> T, \qquad (3)$$

The Huber function ρ(x) is a convex function that has a quadratic form for values less than or equal to a threshold T and a linear growth for values greater than T. The Gibbs PDF of the Huber function is heavier in the tails than a Gaussian. Consequently, edges in the frames are less penalized with this prior than with a Gaussian (quadratic) prior.

To minimize the cost function in, we use the conjugate gradient (CG) iterative method because of its simplicity and efficiency. Compared to some other iterative methods such as Gauss-Seidel (GS) or SOR that need explicit derivation of matrix A when solving a linear equation Ax = b, CG can decompose the matrix A to concatenation of filtering and weighting operations. However, CG can only be used with linear equation sets, whereas the cost function is nonquadratic and so its derivative is nonlinear. To overcome this limitation, we use lagged diffusivity fixed-point (FP) iterative method to lag the diffusive term by one iteration. Using this method for a sample vector x, at thenth iteration the non-quadratic Huber-norm ρ(xn)1 is replaced by the following quadratic form:

$$\mathbf{||\ \rho(x^n)||=(x^n)^T\ V^n(x^n)=||x^n||^2{}_{Vn}} \qquad \textbf{(4)}$$

where Vn is the following diagonal matrix.

C. Blur Estimation:

Within an image or video frame, non-edgeregionsand weak structures are not appropriate for blur estimation. Hence, more accurate results would be obtained if the estimation is not performed in such regions. For this reason the user should first manually select a region with rich edge structure, whereas the most salient edges are automatically chosen. Moreover, sharpening salient edges would also improve the accuracy of blur estimation. The authors of leveraged these two strategies by preprocessing blurred images with the shock filtering method proposed in. Shock filtering is an edge preserving smoothing operation by which soft edges gradually approach step edges within a few iterations while non-edge regions are smoothed. Since shock filtering is sensitive to noise, sometimes a prefiltering operation is applied to first suppress noise. For example, in bilateral filtering (proposed by) is used and a lowpass Gaussian filtering is utilized before shock filtering.A similar conceptforthe blur estimation is exploited in which the image is first sharpened by redistributing the pixels along the edge profiles in such a way that antialiased step edges are produced. Having the sharpened image and the blurry input image, the blur is then estimated using a maximum a posteriori (MAP) framework.

D. FINAL *HR FRAME ESTIMATION*

After the PSF estimation is completed, the final HR frames are reconstructed through minimizing the cost function.

VII. CONCLUSION

A method for blind deconvolution and super resolution from one low-resolution video is introduced in this paper. The complicated nature of motion fields in real-life videos make the frame and blur estimations a challenging problem. To estimate the blur(s), the input frames are first upsampled using non-uniform interpolation (NUI) SR method assuming that the blurs are either identical or have slow variations over time. Then the blurs are determined iteratively from some enhanced edges in the upsampled frames. After completion of blur estimation, the reconstructed frames are discarded and a non-blind iterative SR process is performed to obtain the final reconstructed frames using the estimated blur(s). A masking operation is applied during each iteration of the final frame reconstruction to successively suppress artifacts resulted by inaccurate motion estimation. Comparison is made with the state of the art and the superior performance of our proposed method is confirmed through different experiments.

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EXPERIMENTAL ANALYSIS OF MEDICAL IMAGE CLASSIFICATION AND RETRIEVAL TECHNIQUES

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***Abstract*— Medical Image classification and similar image retrieval are the two important processes in diagnosis and automatic annotation. These help the doctors and radiologists in their decision making during decease identification and decision making. Image classification is usually done by checking its content similarity. Image content is its visual features referring to mathematical attributes. Similarity checking is done by using similarity or dissimilarity measures which are also known as distance metrics. As image attributes are wide in range, the similarity measure worked well for one feature set may not show the similar performance for other. For this reason in this paper we explored various existing similarity measures viz. Manhattan, Cosine, Chi-square and Cramer distances and their effect with respect to image intensity features and wavelet based texture features. We drawn certain conclusions on the performance of these distance metrics in classification and retrieval of IRMA data sets. Mean Average Precision and Average Recall Rates are used in analyzing retrieval performance for analyzing the medical image retrieval and classification task.**

Keywords—medical images, classifiers, retrieval, distance metrics.

I. INTRODUCTION

Image attributes mostly classified as Color, texture and shape. Most of the medical imaging modalities such as XRAY, CT and MRI produced medical images are usually in gray tone. Therefore color features like color histograms and color correlograms would not work well in analyzing visual content in medical images. Texture features are more powerful features in analyzing the medical images. Knowledge extractable from medical images is not precise. Spatial data in the image is not expressed in conventional languages. Most of the image has geometrical information and Medical images arising from photography (e.g., endoscopy, histology, dermatology), radiographic projection (e.g., x-rays, some nuclear medicine), and tomography (e.g., CT, MRI, ultrasound) impose unique, image-dependent restrictions on the nature of features available for abstraction. Similarity from a medical perspective is predominantly context dependent [6].

Different modality based medical images have different characteristics such as Microscopic histology images posses unique color signatures and cell textures. Ultrasound images of large organs appear to be dominated by textures, hence emphasize extracting a global property rather than local features, Image arising from a projection technique, from a tomographic technique such as MRI, Tomographic images grouped by acquisition from individual subjects also have the unique virtue of retaining the data required for un ambiguous 3-D reconstruction of tissue structures Shape matching similarity operations are used here.

Chest X ray images are projections of many overlapping structures. Indexing procedures address textures based features rather shape. Tomographic images readily permit non overlapping geometrically bounded organs and tissues to be identified as a collection of individual features [6]. Texture contains important information about the structural arrangements of their surfaces and their relationships to the surrounding environment. Small area of patches with little gray variation is called TONE. Small area of patches with large gray variation is called TEXTURE. GLCM Features computed for various angular relationships with the distance between pixels. Four 4 x 4 matrices are generated as f(i,j,d,0^0), f(i,j,d,45^0), f(i,j,d,90^0), f(i,j,d,135^0). 'd' varies from 1:n. [1]. Later Tamura at el approximated six textural features contrast, coarseness, directionality, line likeliness, regularity and roughness, which works similar to psycho visual (human visual) perception. Contrast, Coarseness and directionality considered as most powerful features among all texture attributes of an image discussed in [2]. Texture classification and discrimination based on the energies of image subbands using DCT, wavelet and spatial partitioning detailed in [3].

The wavelet transform involves filtering and sub sampling. Compared non orthogonal (Gabor), orthogonal and biorthogonal (tree structured) decompositions wavelet transform decompositions to analyze texture classification [4]. Mean and standard deviation of Daubechies based wavelet coefficients are used as features. Quad tree indexing, Energy estimation (mean and variance) in subbands are used for feature extraction from subbands. Images decomposed into blocks and wavelet coefficients are computed for each block and the query features compared with these features for fast retrieval of similar images from large databases discussed in [5].

Transformation based texture calculation based on energy of subbands using DCT presented in [15]. Statistical textural features obtained with the help of discrete wavelet transforms presented in [16]. Gabor wavelet texture features compared with orthogonal wavelet texture features in [17]. In Operator or pattern based texture calculations, Local Binary Patterns (LBP), Local Ternary Patterns (LTP) and Local Derivative Patterns (LDP) are used. These are more powerful texture features as they are invariant to rotation and scaling proposed in [18].

Color and texture features were compared with nine different similarity measures including Heuristic histogram distances (Minkowski form, Histogram intersection, Weighted mean variance), Non parametric statistical form (Kolmogorov Smirnov, Cramer von Mises, Chi-square), Information theoretic divergences (Kullback - Leibler divergence, Jeffery divergence), ground distance measures (Earthmover's and quadratic distance) in [7],[10]. The selection of a similarity measure substantially improves the efficiency of classifier or retrieval. Histogram quadratic distance incorporates cross bin information via a similarity matrix A. Earth Movers Distance computations complexity is highest. It is minimum cost of transferring one distribution with the other. EMD can be defined as a solution of the transportation problem. Jeffrey divergence is the symmetric version of KL divergence. Chi square measures line likeliness from one distribution being drawn from the other. Perform better with large number of observations. Cramer von Mises used for judging the goodness of a fit of a cumulative distribution functions when compared with empirical distribution function. Information Theoretic (How compact one distribution can be coded using the other one as codebook) Kullback Leibler Divergence (KLD) and Jeffrey Divergence (JD) are the examples of information theoretic. Statistical distances used to test the hypothesis that two empirical distributions that have been generated from the same underlying true distribution. These include Kolmogorov-Smirnov Divergence (KSD), Cramer-von Mises (CvM), Chi Square and Pearson Correlation Coefficient. Ground based distances perceptually meaningful distance measures between individual features. These include Quadratic distance and Earth Mover’s Distance (EMD). Manhattan is best among Minkowski distance metrics. JD is stable over KLD. For large samples chi-square and statistical distances worked better [7], [10].

II. IRMA DATABASE

IRMA image data sets and retrieval of similar modality images based on prior learning of classifier presented by RWTH Aachen university developers, Mono hierarchical multi axial classification code is presented in [8]. 14 digit code is used in IRMA database with first five digits represent TECHNICAL CODE relating medical imaging modality with technical parameters in which digit 1 indicate imaging physical technique (X-ray, US, CT), digit 2 represent modality position (plain projection, fluoroscopy, angiography etc), digit 3 denote technique (digital, analog etc), digit 4 for assessing sub techniques (high energy, low energy, parallel beam, etc) and digit 5 for external ads, drugs and additional markers. Next three digits indicate directional code for orientation of the image with respect to body Digit1: common orientation (coronal, sagital, transversal etc). Digit 2: specific orientation (posterior - anterior, etc) and Digit 3: functional orientation (standing, sitting, inclination) etc. Next 3 digits ANTOMICAL CODE : for body region examined Nine major anatomical regions are extracted. (1. Total body, head/skull, spine etc) and last 3 digits: BIOLOGICAL CODE for biological system under evaluation. (The top level code ten organ systems are specified like cerebral spinal, cardiovascular, respiratory, and gastrointestinal and so on. The code axis is orthogonal and each axis is built mono hierarchically [8].

Assert system analysis is CT images of the lung with respect to 8 certain diagnostic inquires KMeD and COBRA retrieve ventricular shapes extracted from MR images of the head.

I Browse operate on histological slices Interpretation of medical images is dependent on both image and query context.

I-Browse is a CBIR system that integrates iconic and semantic features for histological image analysis and also to do textual annotations for unknown images. Coarse feature detectors use color and gray level histograms and semi-fine feature detection is done by Gabor filter [9].

IRMA concept is based on a conceptual and algorithmic separation of seven processing steps: 1. Categorization using global features 2. Determination of parameters for registration geometry and contrast for each likely category. 3. Feature extraction using local features, Feature selection and combination with respect to category and query content. 4. Indexing resulting in a hierarchical multi scale blob representation and registration. 5. Identification of blobs linking a priori knowledge to image content. 6. Identification of blobs by finding priori knowledge to image content. 7. Retrieval processed on the abstract blob level [8].

III. DISTANCE MEASURES

Minkowski distance metric at different levels presented in [14]. These metrics are preferred when each dimension holds equal importance in retrieval process. Minkowski metric was used for feature vector comparison in [19]. Manhattan distance or City block distance or Minkowski L_1 depends on the rotation of the coordinate system. Cosine Angle Distance (CAD), Chi-square and Mahalanobis distance measures for shape databases evaluated in [20]. According to [21] and [22] the discrete, three dimensional analogous of the Kolmogorov-Simirnov test L_∞, Cramer -von Mises test (L_2) and the Earth Mover Distance (L_1) are the statistical distance metrics.

A. Minkowski Distance

The Minkowski distance of order *p* between two points X and Y belongs to feature space R, $X = (x_1, x_2, x_3..x_i....x_n)$ and $Y = (y_1, y_2, y_3..y_i.....y_n)$ € R is given in equation (1)

$$d(x,y) = \sum_{i=1}^{n}((|x_i - y_i|)^p)^{\frac{1}{p}} \tag{1}$$

In which p represent the order of the distance metric. When p < 1, it is not considered as a distance metric as it does not satisfy the triangular inequality. When p = 1, the distance turns into Manhattan distance and when p = 2 it become Euclidean distance. In the limiting case of *p* reaching infinity, the Chebyshev distance is obtained. Minkowski distance metric is generalized form of Euclidean distance and city block distances, preferred when each dimension holds equal importance in retrieval process.

B. Manhattan or City Block Distance

This metric0 measures the direct grid distance along the pixels and diagonal movements not allowed. Manhattan distance metric retrieve images at a faster rate when compared with Euclidean distance [10]. The metric shown good MAP in both feature similarity measures, but worked well for gray histogram comparison over texture feature similarity. This distance is shown in equation (2).

$$d(x,y) = \sum_{i=1}^{n}|x_i - y_i| \tag{2}$$

C. Euclidean Distance

This distance represent length of the line segment connecting two points in a feature set. In image processing and retrieval, the images are with n dimensional feature vectors and hence n-dimensional Euclidean distance is used. If x and y represent two images then the Euclidean distance between them obtained as shown in equation (3).

$$d(x,y) = \sum_{i=1}^{n}\sqrt{(x_i - y_i)^2} \tag{3}$$

This distance also termed as quadratic distance obtained by using the form as shown in equation (4).

$$d(x,y) = \sqrt{((X-Y)' \cdot (X-Y))^2} \tag{4}$$

In this paper, we used the Euclidean distance for gray intensity and texture feature comparisons.

D. Chebyshev / Chessboard or Infinity Distance

This distance also known as chessboard obtained when limiting value reaches to infinity. This distance between two points (x, y) is expressed as shown in equation (4).

$$d(x,y) = max_{i=1}^{n}|x_i - y_i| \quad (4)$$

E. Cosine Angle Distance

Cosine Angle Distance (CAD) does not follow the triangular similarity. The cosine distance metric normalizes all feature vectors to unit length and makes it invariant against relative in-plane scaling transformation of the image content. This measure is best suited to find the orthogonality between two vectors. If the cosine angle computed between the Eigen values of vectors, it works much better. CAD is shown in equation (5).

$$d(x,y) = \frac{\sum_i x_i y_i}{\sqrt{\sum_i x_i^2 \sum_i y_i^2}} = \frac{x_i \cdot y_i}{||x_i|| ||y_i||} \quad (5)$$

F. Chi-Square Distance:

The chi-square distance measure is used in correspondence analysis and related ordination techniques. Chi-squared distance does not reach a constant, maximal value for sample pairs with no species in common, but fluctuates according to variations in the representation of species with high or low total abundances. Chi-squared tests are often constructed from a sum of squared errors, or through the sample variance as shown in equation (6).

$$d_{chisquare}(x,y) = \sum_{i=1}^{N} \frac{(x_i - \mu_i)^2}{\mu_i} \quad (6)$$

G. Kullback-Leibler Distance

Solomon Kullback and Richard Leibler introduced Kullback Leibler divergence in 1951[38]. It does not obey triangular inequality hence it is not a valid distance metric. This also considered as relative entropy of two distributions and is shown in equation (7).

$$d_{kullback}(x,y) = \sum_{i=1}^{N} x_i log\frac{y_i}{x_i} \quad (7)$$

H. Jeffrey Distance

Jeffrey Divergence is the symmetric version of Kullback-Leibler distance with respect to samples x and y. The divergence equation is shown in equation (8).

$$d_{jeffray}(x,y) = \sum_{i=1}^{N} x_i log\frac{y_i}{\mu_i} + y_i log\frac{x_i}{\mu_i} \quad (8)$$

I. Kolmogorov Smirnov Divergence

This is a non-parametric test and calculates distance between empirical distribution function of the sample with cumulative distribution function of the reference. KS distance shown in equation (9).

$$D_{L_\infty}(x,y) = max_{i=1}^{N}|(X_n - Y_n)| \quad (9)$$

J. Cramer von Mises Divergence

This metric tests the goodness of fit for cumulative distribution function and to compare two empirical distributions. Harald Cramer and Richard Edler von Mises propose this metric [12] [13]. It is an alternative for Kolmogorov Smirnov test.

$$D_{L_1}(x,y) = \sum_{i=1}^{N} |(X_i - Y_i)| \quad (10)$$

K. Earth Mover's Distance

It is a measure of distance between two probability distributions. It works to find the minimum distance function. EMD is shown in equation (11).

$$D_{L_2}(x,y) = \sum_{i=1}^{N} \sqrt{(X_i - Y_i)^2} \quad (11)$$

III. METHODOLOGY

In this paper we presented image retrieval task based on gray histogram features and Haar wavelet based texture features for analyzing the performance of standard geometrical, statistical and cumulative distance measures consisting Euclidean, Manhattan, Chebyshev, Cosine angle, Chi-square, Kullback-Leibler, Jeffrey, Kolmogorov-Smirnov, Cramer von Mises and Earthmover's distances. Experimentation is done in Matlab to analyze the effect of distance metrics on different types of images.

1. Image database is loaded into Matlab workspace.
2. Query Image is selected from the database.

3. Intensity based feature extraction is done by computing gray histograms for query and database images.
4. Using Discrete Haar Wavelet transform (DWT), texture features extracted.
5. Geometrical distance measure Manhattan, statistical distance metrics Chi-square, cumulative statistical measure Cramer and Cosine angle distance were applied for feature similarity measurement.
6. Performance measures, precision (P) as given in eq.13, Mean average precision MAP and recall (R) as given in eq.12. are evaluated for the retrieved images.
7. Comparative analysis is done for image retrieval based on effect of similarity measures for gray intensity and texture features

Precision and recall are used as measurements of classification in CBIR of medical images which are defined as

$$\text{Recall} = \frac{\text{Total number of relevant images retrieved}}{\text{Total relevant images in the data base}} \quad (12)$$

$$\text{Precision} = \frac{\text{Total number of relevant images retrieved}}{\text{Total number of retrieved images}} \quad (13)$$

IV. EXPERIMENTS & RESULTS

We did experimentation of our method on IRMA 2007 and 2008 datasets for classification and retrieval of twenty different groups of anatomical images including lungs, spine, hands, legs ankle, arms and skull images. Sample query images are shown in Fig 1. We presented the comparative analysis of which distance metric performed well for a particular type of query for its gray and texture features comparison. Gray features computed from 64-bin histogram and Discrete Wavelet coefficients computed to represent texture features.

In this paper we experimented on four distance measures Manhattan, Cosine Similarity, Chi-square and Cramer chosen each from the category of geometrical distances, statistical distances and cumulative distances. We compared these distances in terms of Mean Average Precision (MAP) and Average Retrieval Rate (ARR). Among these distances Manhattan distance and Cosine angle distances were shown outstanding performance in terms of MAP and ARR. We also analyzed that wavelet coefficients representation of image features worked in much better way in image classification as well as retrieval when compared with computation of conventional gray scale histograms to represent images.

We also understood from the analysis that medical anatomical images are rich in texture details and the feature set that worked well for one anatomical structure may not show the same effect on other. As an example from Table1 it is clearly shown that for knee and skull images cosine angle distance shown superior performance where as for lungs, spine, neck and hand images Manhattan shown better retrieval compared to others. In terms feature sets, for feet and knee images intensity features worked well over texture features in classification and retrieval. We selected 1000 images from IRMA dataset consisting 50 images of twenty different categories. Selected 5 images of each category as queries and tested the system. Average precision and Recall rate computed for retrieval up to 50 images in terms of each distance metric.

Fig.1. Sample Query Images

Through this experimentation it is observed that Manhattan and Cosine angle distances performed well with DWT features with among all other distance metrics as shown in table I. Image retrieval performed using gray histograms shown in

Fig.2. Retrieval of spine and lung images using wavelet texture features shown in fig.3 and fig.4.

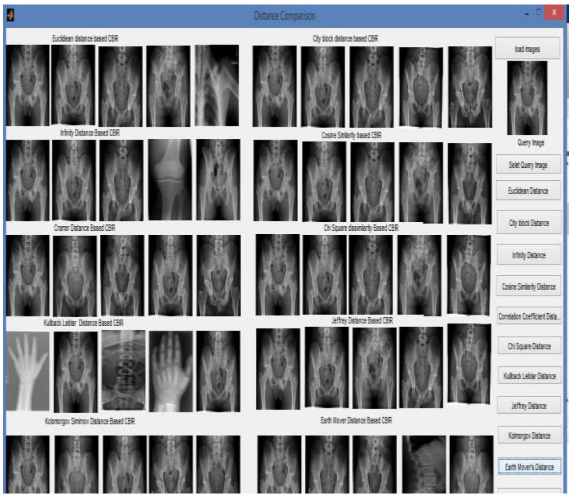

Fig.2. Abdomen images with gray intensity features

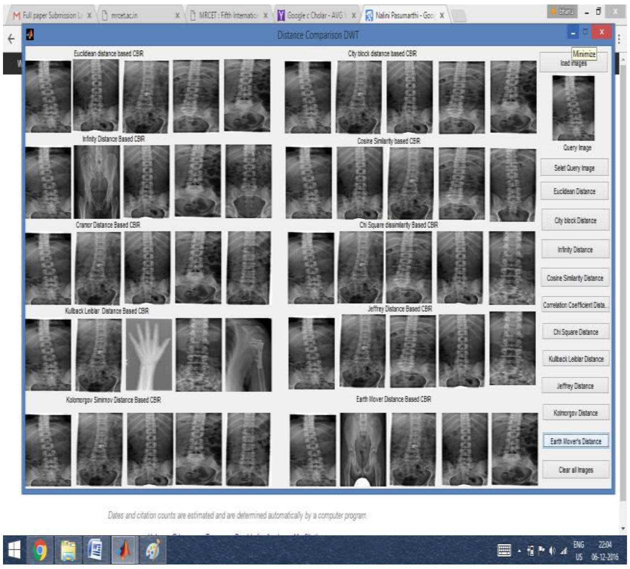

Fig.3. Spine images with DWT texture features

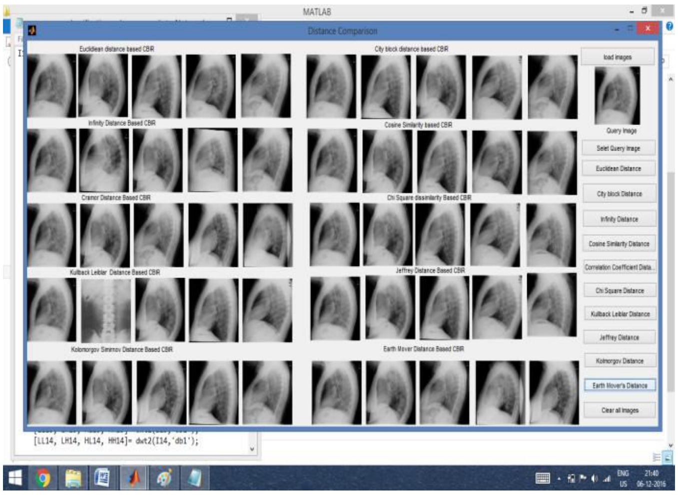

Fig.4. Lung images with DWT texture features

TABLE I Precision and MAP and ARR for intensity and wavelet features

Class of Images	**Manhattan**		**Cosine Angle**		**Chi Square**		**Cramer von Mises**	
	DWT	**HG**	**DWT**	**HG**	**DWT**	**HG**	**DWT**	**HG**
Abdomen	**0.96**	0.58	0.86	0.51	**0.87**	0.63	0.37	0.46
Spine	**0.80**	0.71	0.77	0.66	**0.80**	0.7	0.31	0.54
Lungs (front view)	**0.93**	0.49	**0.95**	0.48	0.83	0.48	0.45	0.45

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Lungs (side View)	**0.94**	0.86	0.86	0.56	**0.87**	0.86	0.38	0.68
Mammogram	**0.67**	0.35	0.5	0.35	0.52	0.33	**0.63**	0.45
Neck	**0.36**	0.36	0.3	**0.36**	0.28	0.22	0.1	0.29
Hand Fingers	**0.38**	0.24	**0.35**	0.26	0.31	0.29	0.19	0.27
Wrist	**0.50**	0.49	0.42	0.28	0.42	**0.45**	0.2	0.34
Hands	**0.45**	0.12	**0.46**	0.11	0.29	0.12	0.12	0.11
Elbow	**0.39**	0.17	**0.33**	0.18	0.35	0.15	0.11	0.15
Feet	**0.49**	**0.39**	0.41	0.29	0.34	0.31	0.08	0.29
Ankle	**0.38**	0.37	0.28	**0.36**	0.35	0.4	0.15	0.34
Knee (side view)	0.31	**0.44**	0.33	0.32	**0.71**	0.43	0.18	0.33
Knee (front view)	**0.73**	0.34	**0.71**	0.29	0.59	0.34	0.16	0.29
Knee (top view)	0.57	0.62	**0.76**	0.63	0.39	**0.64**	0.1	0.37
Shoulder	**0.87**	0.29	0.5	0.18	**0.88**	0.33	0.19	0.22
Skull (front view)	0.48	0.69	**0.89**	0.55	0.38	**0.73**	0.32	0.46
Skull (Side view)	**0.65**	0.56	0.37	0.46	**0.58**	0.51	0.11	0.42
Skull (top view)	**0.39**	0.37	**0.47**	0.29	0.36	0.39	0.23	0.29
Teeth	**0.45**	0.35	**0.46**	0.28	0.23	0.35	0.14	0.27
MAP	**0.59**	**0.44**	**0.55**	**0.37**	**0.52**	**0.43**	**0.23**	**0.35**

VI CONCLUSIONS

In this paper, first we presented an overview of geometric and statistical distance metrics used in image retrieval applications and performed the comparative analysis of four diversified distance measures including Manhattan, Cosine Angle, Chi-square, Cramer von Mises and Earthmover's distance on intensity and texture features. Intensity features extracted by computing gray level 64-bin histograms and texture features through wavelet decompositions. Among geometrical distances Manhattan distance shown outstanding performance and in statistical distance metrics Cosine Similarity, shown good MAP score for all the types of queries. Finally it is observed that geometrical and statistical distance measures - Manhattan, Cosine Angle shown good MAP. Through this experimentation we draw a conclusion that medical image retrieval and classification not only depends on the image content representation but also on the similarity/ distance measures used.

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A STUDY ON "SCORE AGING CALIBRATION FOR SPEAKER VERIFICATION"

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***Abstract*—the time interval between comparison samples as additional information. Several functions are proposed for the incorporation of this time information, which is viewed as aging information, in a The gradual changes that occur in the human voice due to aging create challenges for speaker verification. This study presents an approach to calibrating the output scores of a speaker verification system using conventional linear score calibration transformation. Experiments are presented on data with short-term aging intervals ranging between 2 months and 3 years, and long-term aging intervals of up to 30 years. The aging calibration proposal is shown to offset the decreased discrimination and calibration performance for both short- and long-term intervals, and to extrapolate well to unseen aging intervals. Relative reductions in C_{r} (cost of log-likelihood ratio) of 1–4% and 10–43% are obtained at short- and long-term intervals, respectively. Assuming that a system has knowledge of the time interval between samples under comparison, this approach represents a straightforward means of compensating for the detrimental impact of aging on speaker verification performance.**

***Index Terms*—Aging, calibration, quality measures, speaker variability, speaker verification.**

I. INTRODUCTION

The time interval between speech enrollment and verification events affects the match score output by a speaker verification system. As the time interval increases, system performance becomes more adversely affected. This effect has been experimentally observed in several studies, and has been alluded to in many more.Recent progress in speaker verification has focused on improving performance in the presence of variability external to the speaker, from sources such as the recording and transmission equipment, the characteristics of the recording environment, and the duration of active speech. Within-speaker sources of variability, such as emotion, stress, vocal effort, health and aging, have received less research attention.

'Vocal aging', or the process of change acting on the voice over a given time interval, occurs due to a combination of stimuli, including: the natural processes of physiological and cognitive aging physical or cognitive conditions affecting the speaker's health and social or environmental factors, such as geographical relocation .

The varying degrees to which these stimuli influence a speaker's voice leads to vocal aging that is speaker-dependent. There are however, some characteristics of vocal aging shared among speakers. Practical evidence for this is the relative success of age estimation from speech. This motivates the application of speaker-independent aging compensation strategies to speaker verification.

Previous proposals to compensate for 'long-term' aging (at time intervals of up to 58 years), have been based on a time- interval-dependent decision threshold and on an aging subspace model both evaluated on the Trinity College Dublin Speaker Aging (TCDSA) corpus within a Gaussian mixture model - universal background model (GMM-UBM) speaker verification framework .

In our recent study, aging information was incorporated into conventional score calibration, via quality measure functions, to compensate for 'short-term' aging (at time intervals of 2-36 months). Calibration experiments were presented using male speakers from the Multi-session Audio Research Project (MARP) corpus. This approach was observed to reduce the error rates of an i-vector speaker verification system.

In this paper, we extend the investigation and evaluation of the approach introduced in calibration experiments are presented on a larger set of male and female short-term aging speakers from the MARP corpus, along with a separate set of long-term aging speakers (at time intervals of up to 30 years) from the TCDSA corpus. In addition, the ability of the proposed calibration functions to extrapolate to unseen time differences is assessed.

The extension to score calibration presented in this study operates in a similar way to the time-interval-dependent decision threshold proposed in . In that study, however, a decision threshold was optimized at a specific operating point (the half- total error rate), and the discrimination performance alone was considered. Here, since the time interval is integrated into calibration, performance is optimized over all operating points, and both discrimination and calibration performance are considered. Thus, this approach can be considered a generalized version of that in.

This paper is organized as follows: Section II details the speech corpora and the aging variability contained within. Section III describes the speaker verification system used for experiments. Section IV-A gives an overview of conventional score calibration. Section V introduces the extension of score calibration with aging information. Section VI presents an experimental evaluation.

II. CORPORA

Two corpora are used for the experiments in this study: the Multi-session Audio Research Project (MARP) corpus, and the Trinity College Dublin Speaker Aging (TCDSA) corpus

A. MARP

A subset of the MARP corpus, containing conversational speech, is used as a source of short-term aging data. This sub- set is composed of 60 speakers (35 male, 25 female) of US- accented English, who contributed to recording session's every1–2 months over a period of 3 years. Each session consists of a10 minute conversation with another speaker. The level of participation varied, with between 5 and 19 sessions collected for each speaker. A schematic of the data used in this study, showing the session's per-speaker over time. Sessions were recorded in a soundproof booth using headset microphones, with conversational partners separated by a sound- proof glass division. The recording equipment and environment remained consistent across all sessions.

B. TCDSA

A subset of the TCDSA corpus is used as a source of long-term aging data. The subset is composed of 17 speakers (9 male, 8 female) of Irish- and British-accented English, for whom recordings spanning 15–58 years have been collected. These recordings contain a combination of interviews and speeches, and were sourced exclusively from historical and contemporary radio and television studio broadcasts in the U.K. and Ireland. It is clear that the data available per-speaker over time is variable.

C. Aging Variability

If two recordings of the same speaker are compared in a speaker verification experiment, the time elapsed between the dates of each recording can also be considered. With the MARP corpus, Fig. 1(a), this time difference measurement can be taken as the difference between session indices. For the TCDSA corpus it can be taken as the age difference in year.

To consider all permutations of time difference, an 'all-vs- all' speaker verification evaluation is adopted. In this scheme, each 1 minute recording chunk is treated in turn as a training sample, with all other 1 minute recording chunks serving as testing samples. Same-recording and cross-gender comparisons were omitted. The scores from the resulting trials (comparisons) were analysed according to the *absolute* time difference between their corresponding training and testing chunks. The resulting distribution of target trials (i.e. Same-speaker comparisons) is illustrated for both MARP and TCDSA. The set of possible trials for TCDSA is limited to those with an age difference of $\leq$30 years. The non-target trials (i.e. different-speaker com- persons) are similarly distributed in each case. It is evident that the two corpora offer complementary opportunities to explore short- and long-term aging effects in speaker verification.

D. Trial Weighting

With the all-vs-all approach, the number of possible trials decreases with time difference. Additionally, there is variability in the number of trials across different speakers. This is particularly prevalent in TCDSA where speakers 3 and 16 dominate the target trial distribution.

When evaluating system performance, this implicit bias to- ward shorter time differences, and toward individual speakers, can be suppressed by weighting the scores appropriately. A weighting scheme proposed for speaker recognition evaluations with trials across multiple recording conditions was applied for this purpose. This scheme operates by weighting the contribution of trials of a certain condition by the inverse of the total number of trials for that condition.

In the present case, each speaker-time-difference combination is considered a different ‘condition’. To reduce the number of combinations, individual time differences are grouped into time difference ranges of ≈6 months for MARP and 5 years for TCDSA. The influence of this weighting approach was previously considered for the MARP and TCDSA corpora respectively.

III. SPEAKER VERIFICATION SYSTEM

For the speaker verification experiments, an i-vector framework with probabilistic linear discriminant analysis (PLDA) modelling is used.

At the front end, Mel Frequency Cepstral Coefficients (MFCCs) of 13 dimensions were extracted over 20 ms windows at 10 ms intervals, and appended with first and second or- der derivatives. Frame-based mean and variance normalization and RASTA filtering were subsequently applied. Speech activity detection (SAD) was applied via Combo-SAD, which exploits a combination of voicing and spectral flux features to detect speech in clean and adverse conditions.

A gender-independent Universal Background Model of 1024 components was trained with a subset of NISTSRE 2008 and 2010 data, consisting of approximately 60 hours of microphone speech from US-English speakers, evenly split be- tween male and female speakers. This particular subset was chosen in order to increase the relevance of the UBM to the MARP and TCDSA data. A gender-independent approach was chosen to allow for a subsequent cross-gender score calibration scheme.

A 400-dimensional total variability matrix was estimated from the same recordings used to train the UBM. The i-vectors were post-processed by mean and length normalization, and whitening. To enhance separability, the i-vector dimensionality was reduced to 200 via Linear Discriminant Analysis (LDA). Finally, the within- and between-speaker i-vector variability was modelled with (simplified) PLDA, again using the UBM development data.

The duration of speech for all trials was fixed at 1 minute of active speech (i.e. after applying SAD). Thus, any effects of duration are suppressed.

IV. SCORE CALIBRATION

Score calibration has been recognized as an important component in current speaker verification systems. Calibrated scores can be reliably used across a range of applications, without the need to tune specific decision thresholds. In practical deployment of speaker verification, the calibration of scores is as important as the discrimination ability of the system. In the commonly-used PLDA scoring approach, scores are in theory output as likelihood ratios. However, due to various modelling assumptions, these scores are not true likelihood ratios, and must therefore be calibrated. This is particularly important in the forensic application of speaker verification, so as to avoid misleading interpretations.

Calibration is therefore the procedure of transforming scores from their raw form to calibrated log likelihood ratios (LLRs). In this study, we consider a commonly-employed linear calibration transformation:

$$s = w_0 + w_1 s, \qquad (1)$$

where a raw score s is transformed into a calibrated score s given offset and scaling parameters w_0 and w_1 . These calibration parameters can be acquired by optimization given a set of development data. Logistic regression optimization [32] is used to obtain w_0 and w_1 . To avoid biasing the parameters to- ward individual speakers or time differences, training scores are weighted according to the frequency of their corresponding speaker-time-difference range combination (in the same manner as weighting for performance evaluation; see Section II-D). We refer to the procedure described by Equation (1) as ‘conventional’ score calibration.

A. Conventional Score Calibration:

Conventional score calibration can be approached in several ways when the scores correspond to trials from different conditions. Mandasari proposed three possibilities for score calibration in different duration conditions by varying the number of estimated parameters, as well as the scope of the data used for their estimation. As with our initial study on aging calibra corresponding to trials with an absolute time difference of≤6 months (i.e. ASD = 1–3) are used to train calibration parameters. For TCDSA, the scores corresponding to trials with an absolute time difference of ≤5 years (i.e.AAD = 1–5) are used to train calibration.

B. Conventional Score Calibration Evaluation:

The mismatched, pooled and matched approaches represent various ways in which conventional calibration can be applied to scores with varying time difference 'conditions'. Mismatched performance can be considered as a reference for a naïve system, unoptimized for time differences between trials.

1) Pooled: One set of calibration parameters, w_0 and w_1 is estimated from the pooled scores from trials of all absolute time differences. This approach would be expected to provide better calibration than the mismatched method for time differences>6 months with MARP and >5 years with TCDSA, but potentially worse at smaller time differences, due to generalization of the calibration parameters over all time differences. Note: this pooled approach is equivalent to the 'stacked' approach.

2) Mismatched: One set of calibration parameters, w_0 and w_1, is estimated from the scores of a subset of trials with a small time difference. For MARP, the scores of≤6 months (i.e. ASD = 1–3) are used to train calibration parameters. For TCDSA, the scores corresponding to trials with an absolute time difference of ≤5 years (i.e.AAD = 1–5) are used to train calibration.

V. SCORE AGING CALIBRATION

The variation in score distributions across time difference ranges, suggest that different calibration parameters should be used for each. We propose to extend the conventional score calibration transformation, Equation (1), by incorporating aging information as the time difference between the corresponding trials. The aim is to replace the multiple sets of parameters in matched calibration with a continuous function relating time difference to the linear calibration parameters. We adopt the approach introduced , which incorporated recording duration information in calibration via Quality Measure Functions (QMFs). The extended score calibration is given by:

$$s = w_0 + w_1 s + w_2 Q(\Delta_t), \qquad (2)$$

where Q is a QMF defining the way in which Δ_t (time differ- ence) is incorporated into the calibration. w_2 is a new calibration parameter governing the influence of this term, and is optimized on the development set along with w_0 and w_1. In this study, Δ_t is equal to Absolute Session Difference (ASD) for MARP and Absolute Age Difference (AAD) for TCDSA.

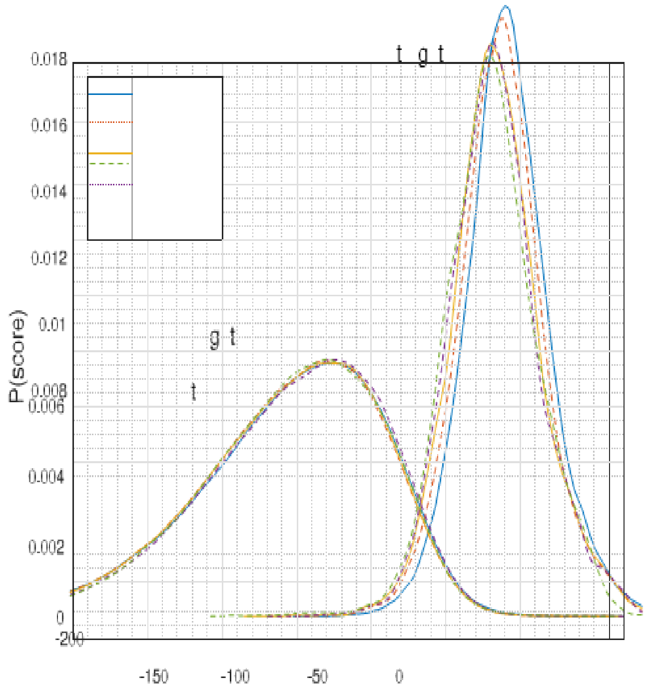
A.Models of Aging Variability:

To investigate the relationship between calibration parameters and time difference range, a shared scaling experiment was applied to the MARP and TCSDSA scores. In this experiment, scores at several different time difference ranges are self-calibrated[2] with the scaling term w_1 held constant (i.e. it is 'shared'), and the offset term w_0 allowed to vary. The change in the offset term across time difference ranges will motivate suitable functions Q to be utilized in Equation (2).

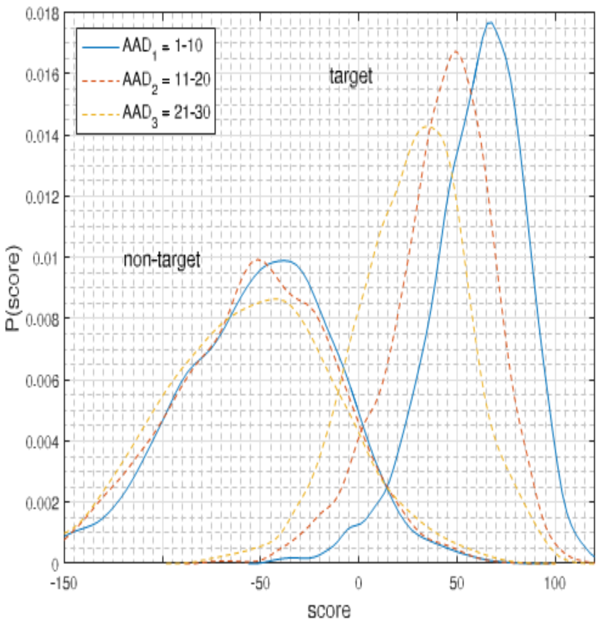
The resulting relationships between offset parameters and time difference ranges for MARP and TCDSA corpora, on a per-gender and combined-gender basis. The trends observed for males and females are generally consistent for both corpora; with MARP, there is an approximately logarithmic increase in w_0 with time difference, and with TCDSA, an approximately linear increase. Thus, both linear and logarithmic functions are considered as suitable candidates for aging calibration. In addition, an exponential function, approximating the w_0 trajectory in Fig. 4(a), is proposed. The candidate QMFs are detailed in Table I. The λ parameter for Q_3 was optimized empirically in our previous study on the MARP corpus . The same value of 0.1 was adopted here, without further optimization.

Due to the relative consistency between genders, we proceed with a calibration experiment involving pooled male and female scores. Based on physiological expectations, and analysis of vocal attributes, differences can be expected in the vocal aging trends of males and females . With a larger speaker pool, it would therefore be preferable to calibrate scores in a gender-specific way.

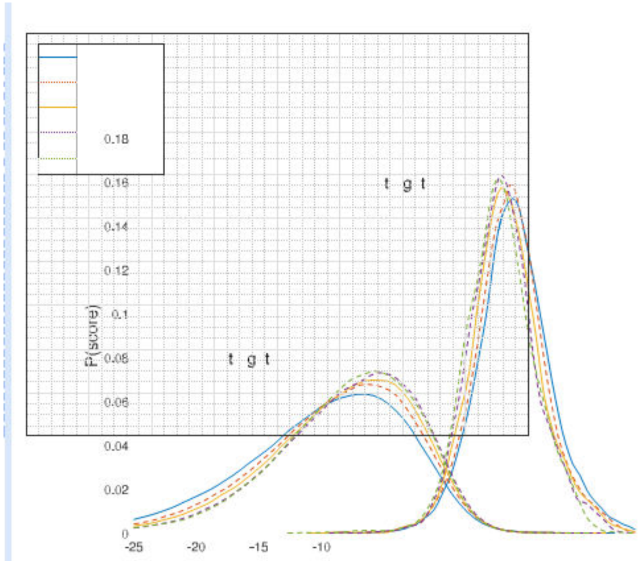
In addition to gender, the absolute age of a speaker affects the extent of aging change; it has been observed that the effect of aging on the voice is non-uniform throughout the adult lifespan, with a greater rate of change typical in those older than 50-60 years of age. Thus, incorporating absolute speaker age into a calibration function would be well motivated. Due to the sparsity of speaker ages in TCDSA and insufficient age

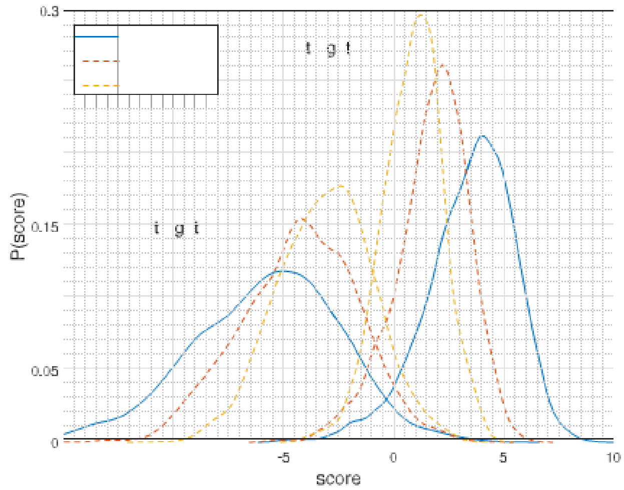


(a)



(b)





(d)

Fig.1. Comparison of score distributions (smoothed using a normal kernel function) before and after 'matched' calibration for several absolute time difference trial subsets. Absolute Session Difference (ASD) and Absolute Age Difference (AAD) units are used for MARP and TCDSA respectively. Solid and dashed lines represent target and non-target distributions respectively

VI. CALIBRATION EXPERIMENTS

Conventional calibration is typically applied by optimizing parameters on a development set and then evaluating on a test set. This procedure assumes that the quantity of data is sufficient for it to be divided into two independent sets with equivalent conditions (distribution of speakers, noise, durations etc.). Given the limited size of the MARP and TCDSA corpora, it is necessary to take a different approach. A cross-validation experiment was therefore applied, whereby the average performance of multiple development and test set divisions is used to compare conventional and score-aging calibration methods. The results are accompanied by appropriate statistical significance tests.

A. MARP Calibration Experiments

For the MARP cross-validation experiment, the 60 available MARP speakers were randomly split into development and test sets in the ratio 5:1. Calibration parameters were estimated from the scores of development speakers, and then used to calibrate the scores of test speakers. There was no overlap between the target or non-target speakers in development and test sets. This procedure was repeated 10 times, with performance metrics evaluated at each iteration. the resulting metrics for each calibration approach, averaged over all 10 iterations. Results for uncalibrated scores have been included for comparison. Relative to the uncal- ibrated case, all calibration methods drastically reduce C_r and R_{mc} . Note that EER and C^{min} are unchanged by conventional mismatched and pooled calibration approaches, which use only score information.

As expected, mismatched calibration performs poorly com- pared to all other calibration approaches; metrics capturing calibration performance, C_r and R_{mc} , are noticeably worse with mismatched calibration. Pooled calibration improves on mismatched calibration across all metrics. All proposed aging calibration functions Q_1 -Q_3 exceed the performance of both mismatched and pooled calibration, and are close to that of matched calibration.

The *p*-values comparing the pairwise C_r of each approach over all 10 iterations is provided. The *p*-values were calculated from a one-sided Wilcoxon signed-rank test . This test is applicable to cross-validation experiments with overlapping folds, where a standard t-test is overconfident. The reduction in C_r with Q_1 -Q_3 , relative to pooled calibration, is statistically significant at a 95% confidence level. In addition, both Q_2 and Q_3 perform better than matched calibration at a significant level. Overall, Q_2 is the most effective function, outperforming all calibration approaches including matched.

In Table IV, the average performance across increasing ASD ranges is presented. Generally, there is a progressive increase in error as ASD range increases; this is anticipated based on the score distributions, and our previous experiments on the MARP corpus. All aging calibration approaches Q_1–Q_3 improve calibration performance relative to pooled calibration, in terms of C_r and R_{mc} , as ASD increases. At ASD_4 and ASD_5 (i.e. for time differences of greater than 2 years), Q_1 –Q_3 exceed both pooled and matched performance. Q_2 is again the most effective function, by a small margin.

B. TCDSA Experiments

Due to the small number of speakers in the TCDSA corpus and the non-uniform distribution of trials across speakers and age differences, a self-calibration experiment with all 17 speakers is presented in addition to a cross-validation experiment.

Performance metrics resulting from the self-calibration experiment, in which the calibration parameters were optimized and evaluated on the same scores, are shown in Table V. As was observed with MARP, all calibration approaches significantly reduce C_r and R_{mc} . The metrics for uncalibrated scores are greater in the case of TCDSA than in MARP (Table II), due to the larger diversity in the TCDSA corpus. Mismatched, pooled and matched calibration result in progressively better performance, which is consistent with MARP. All of the proposed score-aging approaches improve both calibration and discrimination performance relative to all conventional calibration approaches. The performance of Q_1 and Q_2 is equal in terms of C_r , and represents a 7% reduction relative to Pooled calibration.

To establish if the utility of these functions holds when optimized and evaluated on independent data, a cross-validation experiment with a development-test speaker ratio of 12:5 was completed. The resulting average metrics over each of 10 random development-test iterations are presented in Table VI, for the full set of scores, and at three AAD ranges.

Metrics for the full set scores (AAD_{all}) are largely consistent with the corresponding self-calibration is likely caused by the sparsity of trials at each AAD between 1 and 10, see Fig. 2(b). At AAD_2 and AAD_3 , Q_1 –Q_3 exceed both pooled and matched calibration performance across all error metrics.

Improve performance relative to the pooled condition. In general, as more training data is added, the results approach those of the 'full' training condition in the linear ASD/AAD function, improves consistently with added training data for MARP. For TCDSA, the performance with Q1 at the AAD ≤ 20

The approach was evaluated experimentally on two di- verse corpora, containing short-term and long-term aging data.

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A SURVEY ON IMAGE CO-SEGMENTATION VIA SALIENCY CO-FUSION

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Abstract—**Most existing high-performance co-segmentation algorithms are usually complex due to the way of co-labelling a set of images as well as the common need of fine-tuning few parameters for effective co-segmentation. In this paper, gives a survey on the conventional way of co-labelling multiple images, to first exploit inter-image information through co-saliency, and then perform single-image segmentation on each individual image. To make the system robust and to avoid heavy dependence on one single saliency extraction method, to apply multiple existing saliency extraction methods on each image to obtain diverse salient maps. This paper explains the method that fuses the obtained diverse saliency maps by exploiting the inter-image information, which we call saliency co-fusion.**

I. INTRODUCTION

Image co-segmentation refers to the task of extracting common objects from a set of images, which is very useful for many vision and multimedia applications such as object-based image retrieval, image classification, and object recognition. It can be considered as one type of weakly supervised segmentation methods, which makes use of the weak prior that there exist common objects across different images in the set. This is quite different from single image segmentation. The existing single image object-level segmentation methods can only exploit either the prior from human supervision, which requires human interactions such as GrabCut, or the prior from single image-based visual saliency, which might fail at complex images with cluttered background or non-salient foreground. In contrast, image co-segmentation goes beyond single image segmentation in the sense that it can exploit not only the intra-image priors, but also the inter-image priors. Furthermore, it also brings in the new challenges of how to find the right inter-image priors and how to make use of them.

The concept of co-segmentation was first introduced, which used histogram matching to simultaneously segment out the common object from a pair of images. Since then, many co-segmentation algorithms have been proposed in the literature, ranging from early image pair co-segmentation, multiple image co-segmentation , interactive image co-segmentation to the recent multiple objects co-segmentation, multiple group co-segmentation, noisy image set co-segmentation, large-scale co-segmentation , shape alignment targeted co-segmentation and evaluation criteria driven co-segmentation. Despite the great progress made by the existing co-segmentation algorithms, they still have some major limitations. First, most of the state-of-the-art co-segmentation algorithms require fine-tuning of quite a few parameters and the co-labelling of multiple images simultaneously, which are very complex and time-consuming, especially for large diverse datasets. Second, as seen in the existing works, co-segmenting images might not perform better than single image segmentation for some datasets. This might be due to the additional energy term commonly used to enforce inter-image consistency, which often results into unsmooth segmentations in individual mages.

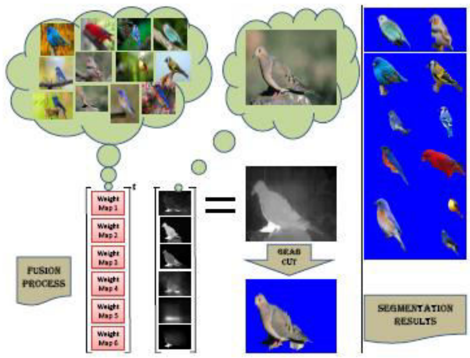

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Fig.1. Fusion of multiple saliency maps of an image generated by different saliency extraction methods to enhance the common foreground object while suppressing background saliency. The fusion process is essentially weighted summation of different saliency maps at superpixel level

In this paper, we focus on binary image co-segmentation, i.e. extracting a common foreground from a given image set. Instead of following the conventional way of co-labelling multiple images, we aim to exploit inter-image information through co-saliency, and then perform single-image segmentation on each individual image. Moreover, to make the system robust and avoid heavy dependence on one single saliency extraction method for generating co-saliency, we propose to apply multiple saliency extraction methods on each image. Eventually, an enhanced saliency map is generated for each image by fusing its various saliency maps via weighted summation at super pixel level, where the weights are optimized by exploiting inter-image information.

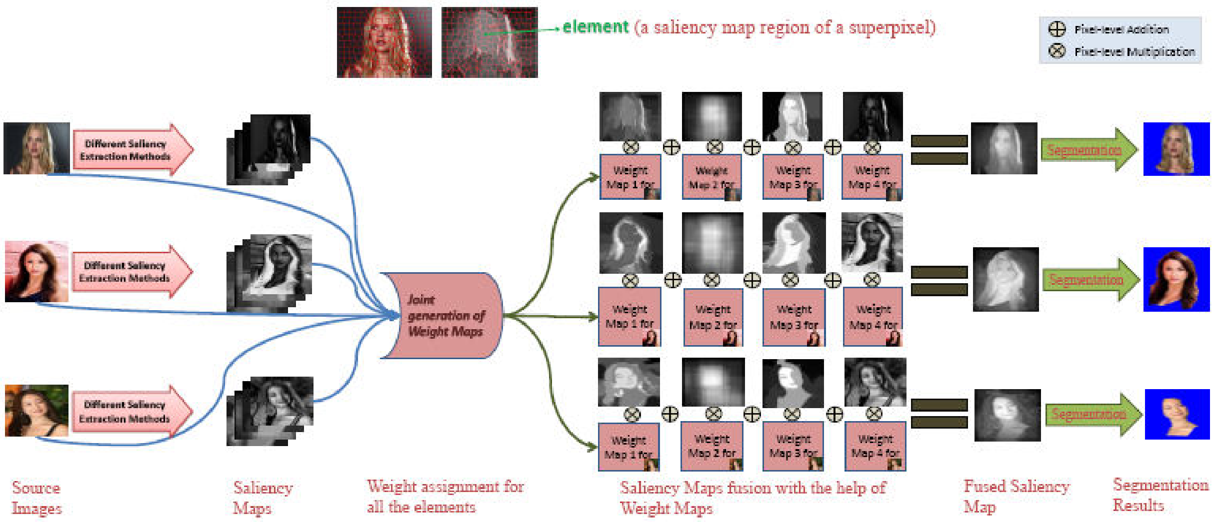

Fig.2. Flowchart of the proposed saliency co-fusion based image co-segmentation where multiple images are used to generate weight maps for fusing different saliency maps of images to extract common foreground. Element, the basic processing unit of our process, is defined as a saliency map region of a super pixel.

Fig. 2 illustrates the process flow of the proposed saliency co-fusion based image co-segmentation. The key component lies in the developed saliency co-fusion process, which is performed at the superpixel level. Particularly, we define each saliency map region (produced by one saliency detection method) of one superpixel as an element (see Fig. 2 top), and give a weight for each element. We formulate the weight selection as an energy minimization problem, where we incorporate saliency recommendations from similar elements, foreground/background priors through similar element voting, and neighbor smoothness constraints. Finally, the fused saliency for a superpixel is just a weighted summation of all the saliency maps of the superpixel. Experimental results show that our saliency co-fusion based co-segmentation achieves competitive performance even without fine-tuning the parameters, i.e., at default setting, compared with the state-of-the-art co-segmentation algorithms. In addition, our by-product, the fused saliency map, exhibits some attractive properties, which could be useful for many other applications.

II. RELATED WORK

Our method is closely related to co-segmentation and co-saliency research.

Co-segmentation: Many co-segmentation algorithms have been proposed in the literature. Early approaches focused on segmenting a pair of images containing one common object. It was later extended to deal with multiple images containing one common object with more effective or more efficient models enforcing inter-image consistency. However, there are also some algorithms being designed for segmenting multiple common foregrounds from a given image.

Our method does not require such human intervention. Recently, applied dense SIFT matching to discover common objects, and co-segment them out from noisy image dataset, where some images do not contain common objects. They tried to enforce inter-image consistency strongly by developing matching based prior, so as to exclude noise image from participating in the co-segmentation process. Co-segmentation was combined with co-sketch for effective co-segmentation by sharing shape templates. Co-segmentation problem was addressed by establishing consistent functional maps across images in a reduced functional space, which requires training. Another interesting work, which reports state-of-the-art performance, employed region level matching. Also, it determined a good co-segment by checking whether it can be well composed from other cosegments. Most of these methods focused on pixel level colabelling whereas we focus on saliency co-fusion. Just like we use multiple saliency maps, there are also some methods that use multiple segmentation proposals to perform semantic segmentation.

For example, made use of multiple segmentation proposals of an image to come up with several compositions and eventually produce semantic segmentation by searching for high-scoring maximal weighted cliques. Later, extended the idea of using multiple segmentation proposals to the object co-segmentation problem and demonstrated better results than classical co-segmentation algorithms. In this research, in contrast to pre-segmenting and then selecting segmentation proposals, we propose to fuse multiple saliency maps to arrive at an enhanced saliency map and then carry out segmentation.

Co-saliency: Co-saliency typically refers to the common saliency existing in a set of images containing similar objects. The term "co-saliency" was first coined in, in the sense of what is unique in a set of similar images, and the concept was later linked to extract common saliency, which is very useful for many practical applications. For example, co-saliency object priors have been efficiently used for co-segmentation. Recently, a cluster based cosaliency method using various cues was proposed, which learns the global correspondence and obtains cluster saliency quite well. It represents state of-the-art method due to its simplicity, effectiveness, and efficiency.

However, the co-saliency method is mainly designed for images of the same object captured at different viewpoints or instances. It cannot well handle image sets with huge intra-class variation. Another recent work fused saliency maps from different images via warping technique and it is able to handle the intraclass variation. However, most of these methods only use a single saliency map which may not be accurate.

III. SALIENCY CO-FUSION

In this section, we first formulate our saliency co-fusion problem. Then we give a detailed description of individual terms as well as implementation details.

A. Problem Formulation Considering a set of N images I = {I1,I2,...,IN}, denote Bn = {Bn 1 ,Bn 2 ,...,Bn M} the set of M saliency maps(normalized to range 0-1) for image In obtained using M different existing saliency extraction methods. Also, denote Pn = {Pn 1 ,Pn 2 ,...,Pn |Pn|} the set of superpixels in image I n obtained using. Defining a saliency map region of superpixel as an element e, which is the basic processing unit in our method, we have total Ne = PN n=1 M|Pn|elements. Let z(n,k,m) denote the associated weight for element e(n,k,m) that belongs to image n, superpixel k, and saliency map m. The weight maps depicted in Fig. 1 and Fig. 2 are basically constructed using these associated weights. We stack all the weights into a vector z = [z1,z2,...,zNe]t for simplicity and use u or v as the element indices for referencing purposes. We mix the usage of the element vector index with its corresponding matrix index (n,k,m) since one can be converted to the other easily. Our goal is to find the optimal weight for each of the elements in order to jointly fuse various saliency maps of similar images at superpixel level such that common foreground saliency gets boosted up and background saliency is suppressed in final fused saliency maps. In particular, we treat saliency co-fusion as a weight selection problem. On one hand, we want to give higher weights to elements with higher confidence. On the other hand, we want to have certain consistency in the weight selection among neighboring elements.

min z Dtz + λztGz s.t. 0 ≤ zu ≤ 1, ∀u ∈ [1,Ne],

M X m=1 z(n,k,m) = 1, ∀In ∈ I,Pn k ∈ Pn (1) where there are two terms traded off by a balancing parameter λ. The first term (Dtz) is a prior term to enforce global commonness and co-saliency, where the prior term coefficient vector D ∈ RNe×1. The second term (ztGz) is a pairwise smoothness term to encourage neighborhood elements to take similar weights, where the smoothness term coefficient matrix G ∈ RNe×Ne. The constraints in Eq. (1) are there to ensure that individual weights range between 0 and 1, and the summation of all the weights for one superpixel is equal to one. Once z is determined by minimizing Eq. (1), the fused saliency map Jn for a pixel p ∈ Pn k can be simply computed as Jn(p) = M X m=1

z(n,k,m)×Bn m(p), (2)

Where Bn m is the m-th saliency map for image.

B. Feature Description and Similarity Unlike other methods where features for matching are extracted from images independent of saliency maps, we develop a saliency map based feature descriptor because our processing units are elements (defined as a saliency map region of a superpixel), instead of pixels or superpixels. We consider the fact that there is no uniformity among saliency maps obtained by different methods. For instance, some saliency maps are of high contrast, while others are of poor contrast. Some are bright, while others are dark. This can cause serious problems in the process if saliency values are directly taken as features. We tackle it by distinguishing potential foreground pixels from potential background pixels in an element using the classical Otsu's method as shown in Fig. 3. For each group (both the potential foreground group and the potential background group in the element), we construct a feature descriptor which consists of the average dense SIFT descriptor, and also the average color values in RGB, HSV, and Lab spaces. However, for each element, we have two feature descriptors with each having dimensions d = 128+3+3+3 = 137. We concatenate them as the feature descriptor for one element. In this way, different elements of the same superpixel obtain different feature descriptors, depending upon the foreground/background distributions in each element.

IV. EXISTING METHOD

This paper is focused on the Co-segmentation problem where the objective is to segment a similar object from a pair of images. The background in the two images may be arbitrary; therefore, simultaneous segmentation of both images must be performed with a requirement that the appearance of the two sets of foreground pixels in the respective images is consistent. Existing approaches cast this problem as a Markov Random Field (MRF) based segmentation of the image pair with a regularized difference of the two histograms - assuming a Gaussian prior on the foreground appearance or by calculating the sum of squared differences. Both are interesting formulations but lead to difficult optimization problems, due to the presence of the second (histogram difference) term. The model proposed here bypasses measurement of the histogram differences in a direct fashion; we show that this enables obtaining efficient solutions to the underlying optimization model. Our new algorithm is similar to the existing methods in spirit, but differs substantially in that it can be solved to optimality in polynomial time using a maximum flow procedure on an appropriately constructed graph. We discuss our ideas and present promising experimental results.

Disadvantages:

- Not Applicable on Stereoscopic Images.
- Less performance ratio on heavy dependence.

V. PROPOSED METHOD

Most existing high-performance co-segmentation algorithms are usually complex due to the way of co-labelling a set of images as well as the common need of fine-tuning few parameters for effective co-segmentation. In this paper, instead of following the conventional way of co-labelling multiple images, we propose to first exploit inter-image information through cosaliency, and then perform single-image segmentation on each individual image. To make the system robust and to avoid heavy dependence on one single saliency extraction method, we propose to apply multiple existing saliency extraction methods on each image to obtain diverse salient maps. Our major contribution lies in the proposed method that fuses the obtained diverse saliency maps by exploiting the inter-image information, which we call saliency co-fusion. Experiments on five benchmark datasets with eight saliency extraction methods show that our saliency co-fusion based approach achieves competitive performance even without parameter fine-tuning when compared with the state-of-the-art methods.

Advantages:

- Applicable on Stereoscopic Images
- Better performance ratio on heavy dependence

Disadvantage:

- Less Usage of deep contrast feature

VI. EXTENSION METHOD

Visual saliency is a fundamental problem in both cognitive and computational sciences, including computer vision. In this paper, we discover that a high-quality visual saliency model can be learned from multi-scale features extracted using deep convolutional neural networks (CNNs), which have had many successes in visual recognition tasks. For learning such saliency models, we introduce a neural network architecture, which has fully connected layers on top of CNNs responsible for feature extraction at three different scales. The penultimate layer of our neural network has been confirmed to be a discriminative high-level feature vector for saliency detection, which we call deep contrast feature. To generate a more robust feature, we integrate handcrafted low-level features with our deep contrast feature.

VII. CONCLUSION

We have studied a novel saliency co-fusion approach for the purpose of image co-segmentation which uses the association of similar images to fuse multiple saliency maps of an image in order to boost up common foreground saliency and suppress background saliency.

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REVIEW OF MODE DECISION IN INTRA FRAME PREDICTION IN AVC/HEVC

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Abstract—video standards such as H.264/AVC are extremely efficient in terms of coding efficiency at the expense of higher computational complexity. HEVC (H.265) standard is the latest enhanced video coding standard which was planned to improve the rendered specifications of its preceding standard MPEG-4 (H.264). According to the H.265 "The main goal of the HEVC standardization effort is to enable significantly improved compression performance relative to existing standards—in the range of 50% bit-rate reduction for equal perceptual video quality"]. In H.264 Intra-frame can make use of either 4x4, 8x8 or 16x16 block sizes for intra-frame prediction. There are nine prediction directions each for 4x4 and 8x8 block size and four prediction directions for 16x16 block size. Intra-picture prediction is a tool in HEVC which "uses some reduction of data spatially from region-to region within a specific picture, but has no dependence on other pictures in the video frames. Although both of them has the same approach in for spatial prediction of pictures based on spatial sample prediction followed by transform coding, H.265 intra-frame prediction uses much more developed features compared to H.264. In this paper the main concentration is the selection of mode for intraframe prediction by using SAD. The mode which is lesser SAD that particular mode is selected for current macro block.

Keywords—Macro block, mode decision, SAD, HEVC, AVC

I. INTRODUCTION

Compression is the basic process of reducing the size of data in order to save storge space and transmission band width. Compression consist of removing redundancies (spatial,spectra and temporal) and encoding the true information in the form of appropriate to suite for applications. There are two compression techniques ie. Lossless and lossy techniques. In lossless the reconstructed image after compression is identical to original image, this method achieve maximum compression ratio. To compress data ,it is important to recognize redundancies in data in the form of coding redundancy, inter- pixel redundancy, and psycho-visual redundancy. Data redundancies occur when unnecessary data is used to represent source information. compression is achieved when one or more of these types of redundancies are reduced.

Video coding has been developed in response to the demands in decreasing bitrates in video broadcasting in order to use communication resources more efficiently. Due to various applications and related data rates many algorithms have been developed. The blocks involving in the video coder algorithms can be shown as in Figure 1.1. This model is based on the assumption about spatial and temporal correlation between pixels and sequences

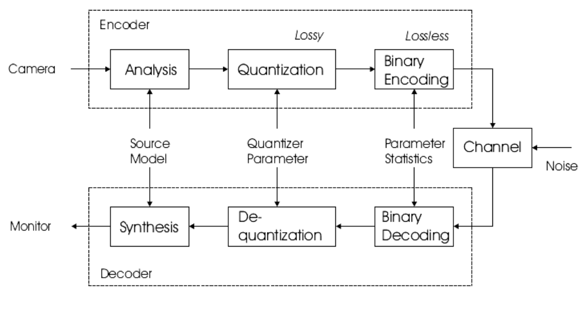

Fig.1. Overview of video coder

II. H.264/AVC

Intra-frame prediction in AVC is done in macro-blocks. For every block in the current frame there is high correlation between samples in the block and the one in the adjacent blocks that makes the adjacent coded blocks as reference to current block. AVC has defined three choices for sizes of the blocks, namely 16x16, 8x8 and 4x4. As it is shown in Figure 2.1 the samples above and to the left named A-M are obtained from previously encoded and reconstructed blocks and are available in encoder and decoder to form prediction references. Figure 2 and Figure 3 shows the Macro block and direction modes for a 4x4 block.

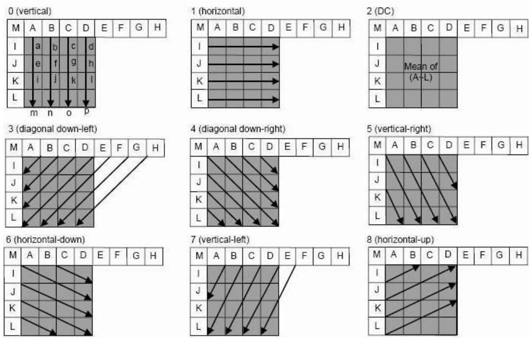

Fig 2. Intra prediction modes[0-8]

M	A	B	C	D	E	F	G	H
I	a	b	c	d				
J	e	f	g	h				
K	i	j	k	l				
L	m	n	o	p				

Fig.3 A 4X4 Macro block

Mode 0(vertical)	Mode 1 (horizontal)	Mode 2 (DC)
▪ a,e,i,m = A ▪ b,f,j,n = B ▪ c,g,k,o = C ▪ d,h,l,p = D	▪ a,b,c,d = I ▪ e,f,g,h = J ▪ i,j,k,l = K ▪ m,n,o,p = L	▪ All the predicted pixels are equal to mean value of all references (A, B...L)
Mode 3 (diagonal down-left)	**Mode 4 (diagonal down-right)**	**Mode 5(vertical-right)**
▪ a = (A+2B+C+2)/4 ▪ b,e = (B+2C+D+2)/4 ▪ c,f,i = (C+2D+E+2)/4 ▪ d,g,j,m = (D+2E+F+2)/4 ▪ h,k,n=(E+2F+G+2)/4 ▪ l,o = (F+2G+H+2)/4 ▪ p = (G+3H+2)/4	▪ d = (B+2C+D+2)/4 ▪ c,h = (A+2B+C+2)/4 ▪ b,g,l = (Q+2A+B+2)/4 ▪ a,f,k,p=(A+2Q+I+2)/4 ▪ e,j,o = (Q+2I+J+2)/4 ▪ i,n = (I+2J+K+2)/4 ▪ m = (J+2K+L+2)/4	▪ a,j = (E+A+1)/2 ▪ b,k = (A+B+1)/2 ▪ c,l = (B+C+1)/2 ▪ d = (C+D+1)/2 ▪ e,n = (I+2E+A+2)/4 ▪ f,o = (E+2A+B+2)/4 ▪ g,p = (A+2B+C+2)/4 ▪ h = (B+2C+D+2)/4 ▪ i = (E+2I+J+1)/4 ▪ m = (I+2J+K+2)/4

Mode 6(horizontal-down)	Mode 7(vertical-left)	Mode 8 (horizontal-up)
▪ a,g = (E+I+1)/2 ▪ b,h = (I+2E+A+2)/4 ▪ c = (E+2A+B+2)/4 ▪ d = (A+2B+C+2)/4 ▪ e,k = (I+J+1)/2 ▪ f,l = (E+2I+J+2)/4 ▪ o,i = (J+K+1)/2 ▪ j,p = (I+2J+K+2)/4 ▪ m = (K+L+1)/2 ▪ n = (J+2K+L+2)/4	▪ a = (A+B+1)/2 ▪ b,i = (B+C+1)/2 ▪ c,j = (C+D+1)/2 ▪ d,k = (D+E+1)/2 ▪ e = (A+2B+C+2)/4 ▪ f,m = (B+2C+D+2)/4 ▪ g,n = (C+2D+E+2)/4 ▪ h,o = (D+2E+F+2)/4 ▪ l = (E+F+1)/2 ▪ p=(E+2F+G+2)/4	▪ a = (I+J+1)/4 ▪ b = (I+2J+K+2)/4 ▪ c,e = (J+K+1)/2 ▪ d,f = (J+2k+L+2)/4 ▪ g,i = (K+L+1)/2 ▪ h,j = (K+2L+L+1)/4 ▪ k,m,l,n,o,p = L

Fig. 4.Modes from 0-8 for a 4x4 block in AVC intra-prediction

In intra mode endcoding , a prediction block is formed based on previously encoded and reconstructed (but **un-filtered**) blocks. This prediction block P is subtracted from the current block prior to encoding. For the luminance (luma) samples, P may be formed for each 4x4 sub-block or for a 16x16 macroblock. There are a total of 9 optional prediction modes for each 4x4 luma block; 4 optional modes for a 16x16 luma block; and one mode that is always applied to each 4x4 chroma block.

Fig5. Original Macro block

The 9 prediction modes (0-8) are calculated for the 4x4 block shown in Figure 5. Figure 4 shows the prediction block P created by each of the predictions. The Sum of Absolute Errors (SAE) for each prediction indicates the magnitude of the prediction error. In this case, the best match to the actual current block is given by mode 7 (vertical-right) because this mode gives the smallest SAE; a visual comparison shows that the P block appears quite similar to the original 4x4 block

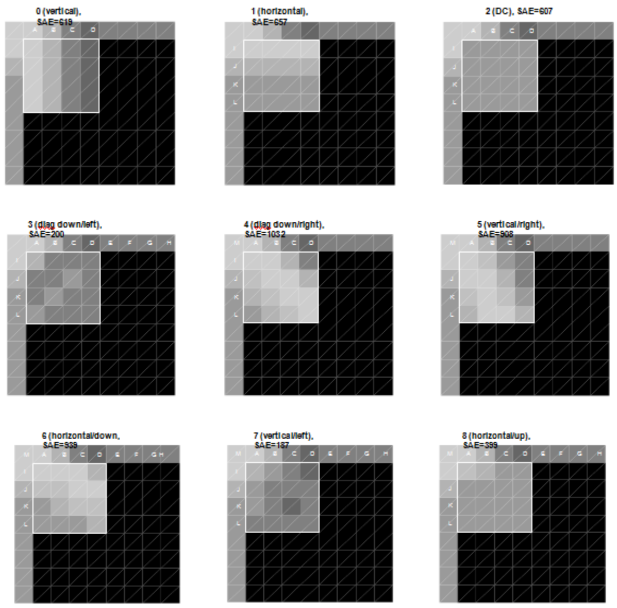

Fig.6. Prediction modes for the block in fig.5

III. H.265/HEVC

Intra-frame prediction in HEVC like AVC is based on prediction of pixels which were decoded in previous blocks and usually the process is followed by a discrete cosine transform (DCT) coding . Many aspects are developed in HEVC in comparison with its ancestor counterpart, among them the first is supported range for block size. In AVC the block sizes exceeds up to 16x16, this size is very small for high definition frame and cannot contain some textures, in HEVC the coding block sizes are supported till 32x32. The second factor to consider is number of angular prediction modes, AVC uses 8 angular, by increasing up to 33 in HEVC it led to increase the accuracy of in directional structure. Like AVC, HEVC utilizes the basic set of samples that are located on top and left of the prediction block. In the current literature reference samples are noted by R(x, y) while the origin of the (x, y) is located on one pixel top and to the left of the block top-left corner. Similarly P(x, y) are the pixels that are to be predicted.

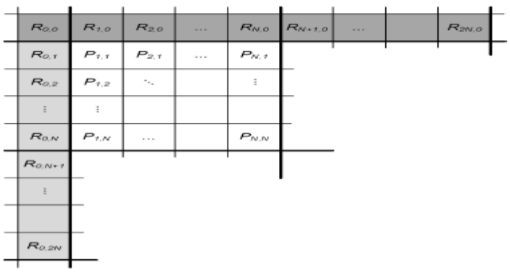

Fig 7. Block size NxN uses R(x, y) as reference in order to predict P(x, y

In the case that in the left column reference is missed it can be compensated by replacing the value of the closest reference in below or in the case of absence that pixel in below the one on the top can be used. Similarly for top reference row the value of the closest pixel from the left can be replaced. It should be noted that if there is no reference sample, a nominal value average of the bit is assigned, for example 128 for 8 bit data. The difference HEVC from AVC in referencing is that HEVC takes advantage of the additional samples in below-left side (R0,N+1 ,...R0,2N+1), while AVC only use left, above and above-right side sample references.

More angular prediction modes (33 Angular modes) compared to H.264 empowered HEVC to have higher coding efficiency in different video materials. However, the number of modes was selected considering the importance of a trade off between coding efficiency and less encoding complexity. Blocks with the size from 4x4 to 32x32 are used in HEVC, and each supports 33 directional modes, which is equal to 132 combinations for block sizes and directional modes.

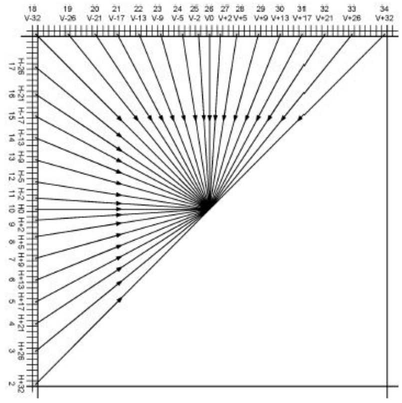

Fig 8. Angular Prediction modes

IV. MODE DECISION

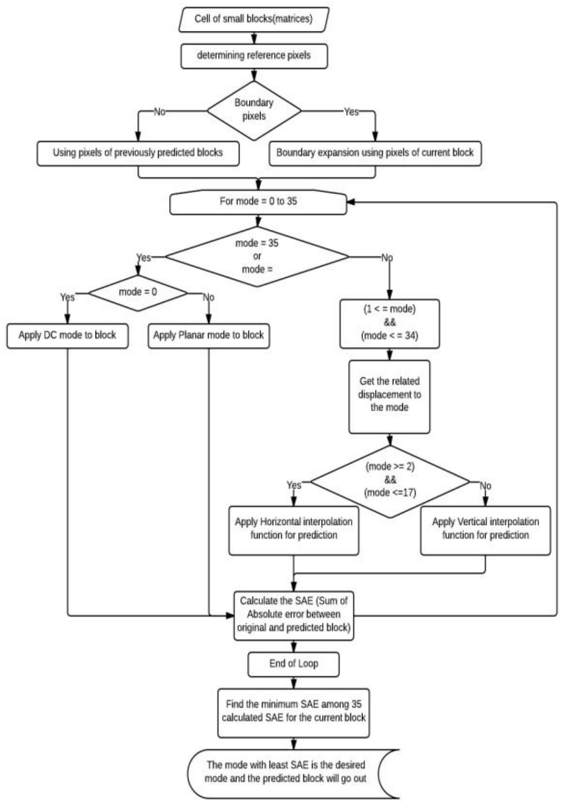

Fig. 9 . *Flow Chart for mode decision*

The main stages of the coded intra-frame prediction function has been illustrated in the flowchart of Fig. 9 Evidently, after the reference samples definition, in a loop, each block would be predicted with all 35 modes and each prediction will be stored in a separate matrix. In the prediction process depending on thecurrent mode in the loop, the vertical prediction function and its related parameters is used for modes between 18 to 34 and horizontal prediction function and its related parameters is uded for modes between 2 to 17. A very crucial point during the angular prediction process is to address properly the projected pixel locations which would have fallen outside the reference sample vector (row vector for vertical and column vector for horizontal prediction)

V. COMPARISION OF INTRA FRAME PREDICTION OF HEVC AND AVC

The figure below shows the comparison of implementation of the two abovementioned algorithms on three different images in VGA, HD-720 and HD-1080 resolutions.

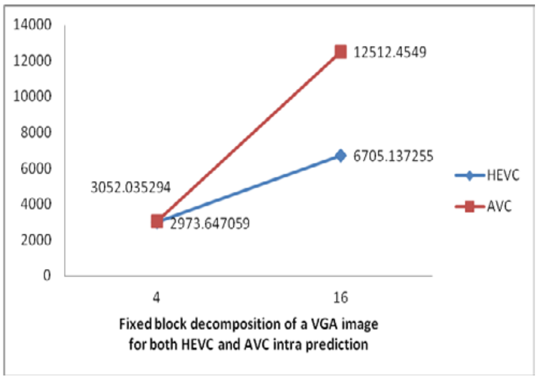

Fig.10. Normalized SAE trend with increase in block size for VGA sample image for HEVC and AVC

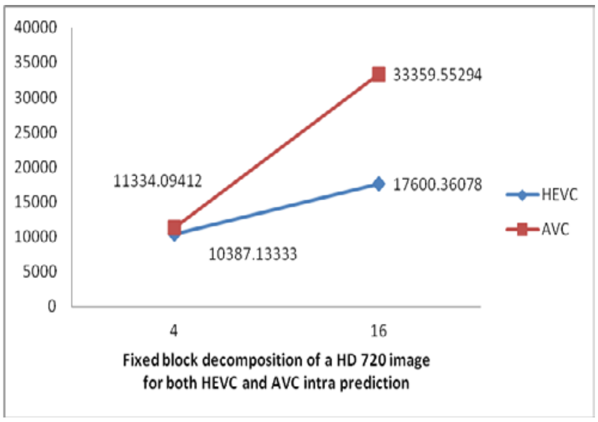

Fig.11. Normalized SAE trend with increase in block

size for HD-720 sample image for HEVC and AVC

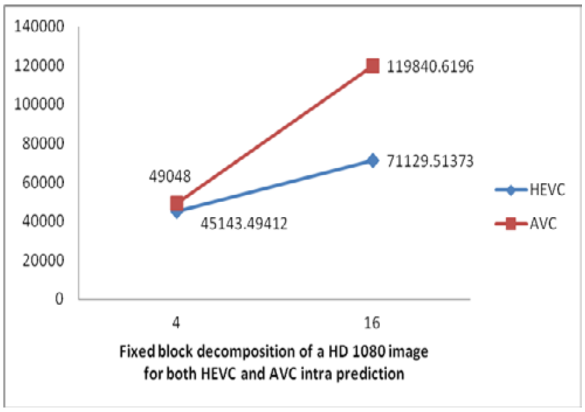

Fig.12. Normalized SAE trend with increase in block size for HD-1080 sample image for HEVC and AVC

Although in all three resolutions, for the smallest block size 4x4, the result of HEVC intra-frame prediction is slightly better than AVC's, in 16x16 blocks it's much more evident that due to higher flexibility and greater optimizations in HEVC intra-frame prediction, the output of the prediction process of HEVC is almost always significantly better than what AVC gives.

VI. CONCLUSION

As it can be seen from the provided figures and graphs, HEVC as the legacy of AVC gives better output quality owing to its flexible algorithm in terms of more angular modes (33 angular modes for HEVC compared to 8 angular modes for AVC intra-frame prediction) and greater versatility in image partitioning or decomposition. It's worth mentioning that outputs of HEVC intra-frame prediction are also completely independent of image resolution and amount of details in the input image.

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AN OVERVIEW ON MILLIMETER WAVE TECHNOLOGY FOR FUTURE WIRELESS COMMUNICATIONS

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Abstract— **Almost all mobile communication systems today use spectrum in the range of 300 MHz–3 GHz. Due to the increasing popularity of smart phones and other mobile data devices such as netbooks and ebook readers, mobile data traffic is experiencing unprecedented growth. In order to meet this exponential growth, improvements in air interface capacity and allocation of new spectrum are of chief important. As the mobile data demand grows, the sub-3 GHz spectrum is becoming increasingly crowded. On the other hand, a vast amount of spectrum in the 3–300 GHz range remains underutilized. The 3–30 GHz spectrum is generally referred to as the super high frequency (SHF) band, while 30–300 GHz is referred to as the extremely high frequency (EHF) or millimeter- wave band. Since radio waves in the SHF and EHF bands share similar propagation characteristics, we refer to 3–300 GHz spectrum collectively as millimeter-wave bands with wavelengths ranging from 1 to 100 mm. The availability of the 60 GHz band as unlicensed spectrum has inspires interest in gigabit-per-second. In this paper, we justify why the wireless community starts looking at the 3–300 GHz spectrum for mobile broadband applications. The applications of mmWave are immense: wireless local and personal area networks in the unlicensed band, 5G cellular systems, vehicular area networks, ad hoc networks, and wearables.**

Keywords—Mm wave, Massive MIMO, 5G.

I. INTRODUCTION

The development of wireless communication in the 60 GHz unlicensed band was the topic of tremendous amounts of research. Much more efforts have been involved in developing more power efficient 60 GHz RFICs. In this article, we explore the 3–300 GHz spectrum and describe a millimeter-wave mobile broadband (MMB) system that utilizes this vast spectrum for mobile communication. MmWave makes use of spectrum from 30 GHz to 300 GHz whereas most consumer wireless systems operate at carrier frequencies below 6 GHz [1] [2] . The main benefit of going to MmWave carrier frequencies is the larger spectral channels, high throughput in small geographical areas, gigabit-per-second data rates ,high bandwidth and much more. MmWave communication could also provide important benefits in other application scenarios like wearable networks, vehicular communications, or autonomous robots. Thus MmWave is receiving tremendous interest by academia, industry, and government for 5G cellular systems. The potential for MmWave is immense. This article is organized by introducing millimeter wave band in chapter1, Challenges for using Mmwave for communications in chapter2, Need of improvement in architecture of cellular communication for using Mmwaves in chapter3, applications of Mmwave communications in chapter 4 and concluding with areas of more research enhancement need to use Mmwaves. Clearly the future is bright for new applications of MmWave.

II. WHY MILIMETER WAVE?

Today's 4G, wireless digital networks made it possible for smartphones and tablets to deliver voice and data communications with bandwidths measuring many millions of bits per second. Specific data speeds vary by carrier. The next generation 5G wireless will have to deliver a huge data rate to handle surging mobile network traffic. According to Cisco Systems' most recent Visual Networking Index (VNI), Global mobile data traffic grew 74 percent in 2015. More than half a billion (563 million) mobile devices and connections were added in 2015.

In 2015, on an average, a smart device generated 14 times more traffic than a nonsmart device. Mobile video traffic accounted for 55 percent of total mobile data traffic in 2015. Globally, 97 million wearable devices (a sub-segment of the machine-to-machine

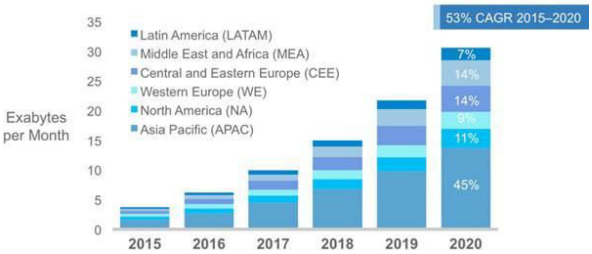
[M2M] category) in 2015 generated 15 petabytes of monthly traffic. [3]

Mobile data traffic will reach the following milestones within the next 5 years:

- Monthly global mobile data traffic will be 30.6 exabytes by 2020.
- The number of mobile-connected devices per capita will reach 1.5 by 2020.
- The average global mobile connection speed will surpass 3 Mbps by 2017.
- The total number of smartphones (including phablets) will be nearly 50 percent of global devices and connections by 2020. Global mobile data traffic will increase nearly eightfold between 2015 and 2020

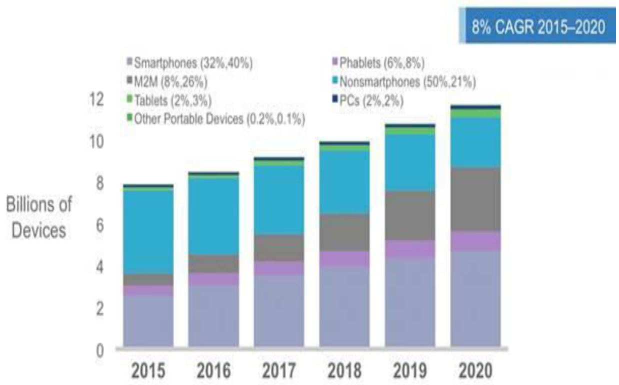
Overall mobile data traffic is expected to grow to 30.6 exabytes per month by 2020, an eightfold increase over 2015.

Figure 1. Global Mobile Data Traffic Forecast by Region



Source: Cisco VNI Mobile, 2016

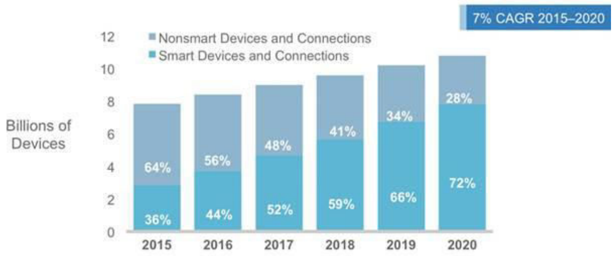
Figure2 .<u>Global Mobile Devices and Connections Growth</u>



Figures in parentheses refer to 2015, 2020 device share.

Source: Cisco VNI Mobile, 2016

Figure 3. Global Growth of Smart Mobile Devices and Connections



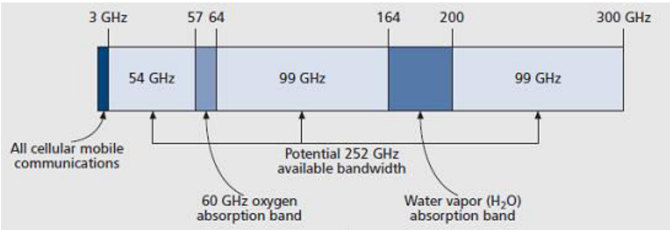
Percentages refer to device and connections share.

Source: Cisco VNI Mobile, 2016

One of the most promising potential 5G technologies under consideration is the use of high-frequency signals in the <u>millimeter-wave frequency band</u> that could allocate more bandwidth to deliver faster, higher-quality video and multimedia content. Millimeter wave (mmWave) cellular systems, operating in the 10-300GHz band, appear to be a promising candidate for next-generation cellular systems by which multiple gigabit-per-second data rates can be supported. Engineers at Samsung estimate that government regulators could free <u>as much as 100 GHz of millimeter-wave spectrum for mobile</u> communications about 200 times what mobile networks use today. Samsung's engineers say their technology can overcome these challenges by using an array of multiple antennas to concentrate radio energy in a narrow, directional beam, thereby increasing gain without upping transmission power. Such beam-forming arrays, long used for radar and space communications, are now being used in more diverse ways.[5]

Recent studies suggest that mm-wave frequencies could be used to enlarge currently saturated 700 MHz to 2.6 GHz radio spectrum bands for wireless communications. The combination of cost-effective CMOS technology that can now operate well into the mm-wave frequency bands, and high-gain, steerable antennas at the mobile and base station, strengthens the viability of mm-wave wireless communications. Further, mm-wave carrier frequencies allow for larger bandwidth allocations, which translate directly to higher data transfer rates. Mm-wave spectrum would allow service providers to significantly expand the channel bandwidths far beyond the present 20 MHz channels used by 4G customers . By increasing the RF channel bandwidth for mobile radio channels, the data capacity is greatly increased. Mm-wave frequencies, due to the much smaller wavelength, may exploit polarization and new spatial processing techniques, such as massive MIMO and adaptive beamforming

Figure 4. Millmeter wave spectrum



UNLEASHING THE 3–300 GHZ SPECTRUM

Almost all commercial radio communications including AM/FM radio, high-definition TV, cellular, satellite communication, GPS, and Wi-Fi have been contained in a narrow band of the RF spectrum in 300 MHz–3 GHz. This band is generally referred to as the *sweet spot* due to its favorable propagation characteristics for commercial wireless applications. The portion of the RF spectrum above 3 GHz, however, has been largely unexploited for commercial wireless applications. Unlicensed use of ultra-wideband (UWB) in the range of 3.1–10.6 GHz frequencies has been proposed to enable high data rate connectivity in personal area networks. The use of the 57–64 GHz oxygen absorption band It is also being promoted to provide multigigabit data rates for short-range connectivity and wireless local area networks. Additionally, local multipoint distribution service (LMDS) operating on frequencies from 28 to 30 GHz was conceived as a broadband, fixed wireless, point-to-multipoint technology for utilization in the last mile. Within the 3–300 GHz spectrum, up to 252 GHz can potentially be suitable for mobile broadband as depicted in Fig.4. Millimeter waves are absorbed by oxygen and water vapor in the atmosphere. The frequencies in the 57–64 GHz oxygen absorption band can experience attenuation of about 15 dB/km as the oxygen molecule (O2) absorbs electromegnetic energy at around 60 GHz. The absorption rate by watervapor (H2O) depends on the amount of watervapor and can be up to tens of dBs in the range of 164–200 GHz . We exclude these bands for mobile broadband applications as the transmission range in these bands will be limited. With a reasonable assumption that 40 percent of the remaining spectrum can be made available over time, millimeter-wave mobile broadband (MMB) opens the door for a possible 100 GHz new spectrum for mobile communication more than 200 times the spectrum currently allocated for this purpose below 3 GHz.

III. CHALLENGES FOR USING MMWAVE FOR COMMUNICATIONS

1. ATMOSPHERIC AND RAIN ABSORPTION

Within the unlicensed 60-GHz band, the absorption due to rain and air particularly the 15 dB/km oxygen absorption are more perceptible. But these absorptions are insignificant for the urban cellular deployments, where base station spacing's might be on the order of 200 m. But actually, these type of absorptions are useful as it will effciently increase the segregation of each cell by further attenuating the background interference from more distant base stations .

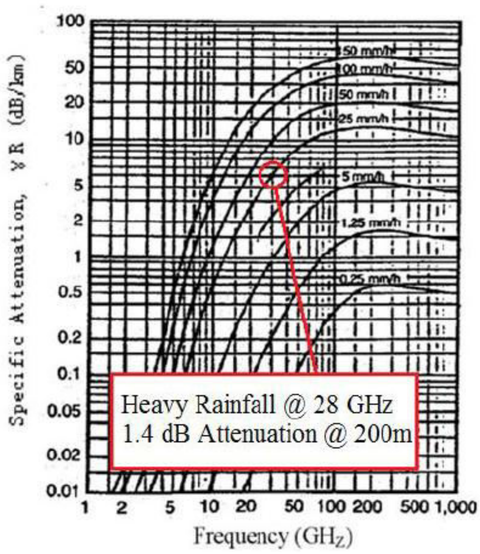
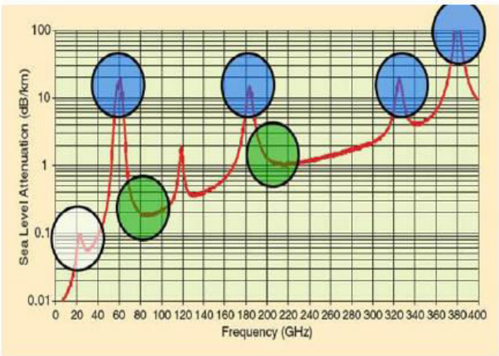


Figure 6. Atmospheric absorbtion across mm-wave in db/km

So from the above explanation, it can be informed that the propagation losses for millimeter wave frequencies are resolvable, but only by steering the beam energy with the help of large antenna arrays and then collect it coherently.Fig.5 and Fig.6 show the rain attenuation and atmospheric absorption characteristics of mm-wave propagation. It can be seen that for cell sizes on the order of 200 m, atmospheric absorption does not create significant additional path loss for mm-waves, particularly at 28 GHz and 38 GHz. Only 7 dB/km of attenuation is expected due to heavy rainfall rates of 1 inch/hr for cellular propagation at 28 GHz, which translates to only 1.4 dB of attenuation over 200 m distance. Work by many researchers has confirmed that for small distances (less than 1 km), rain attenuation will present a minimal effect on the propagation of mm-waves at 28 GHz to 38 GHz for small cells .[4]

Figure 5.Rain attenuation in db/km across frequency at various rainfall rates



2. PATH LOSS

The free space path loss is dependent on the carrier frequency, as the size of the antennas is kept constant which is measured by the wavelength λ=c/*f*, where *f* is the carrier frequency. As the carrier frequency increases, the size of the antennas reduces and their effective aperture increases withthe factor of $\lambda^2/4\pi$, while the free space path loss between a transmitter and a receiver antenna grows with f^2 . So, if carrier frequency *f is chosen* from 3 to 30 GHz, it will correspondingly add 20 dB of power loss irrespective of the transmitter-receiver distance. But for increased frequency, if the antenna aperture at one end of the link is kept constant, then the free-space path loss remains unchanged. Additionally, if both the transmitter and receiver antenna apertures are kept constant, then the free space path loss decreases with f^2

3. BLOCKING

Microwave signals are less prone to blockages but it deteriorates due to diffraction. But, mm wave signals suffer less diffraction than the microwave signals and exhibit specular propagation, which makes them much more vulnerable to blockages. Recent studies reveals that, with the increase in the transmitter and receiver distance the path loss increases to 20 dB/decade under Line of sight propagation, but descents to 40 dB/decade plus an added blocking loss of 15_40 dB fornon-line of sight.

4. PENETRATION AND OTHER LOSSES

For 3–300 GHz frequencies, atmosphere gaseous losses and precipitation attenuation are typically less than a few dB per kilometer , excluding the oxygen and water absorption bands. The loss due to reflection and diffraction depends greatly on the material and the surface. Although reflection and diffraction reduce the range of millimeter- wave, it also facilitates non-line-of-sight (NLOS) communication. While signals at lower frequencies can penetrate more easily through buildings, millimeterwave signals do not penetrate most solid materials very well. The indoor coverage in this case can be provided by other means such as indoor millimeter-wave femtocell or Wi-Fi solutions. It should be noted that next-generation Wi-Fi technology using 60 GHz millimeter waves is already being developed in IEEE 802.11ad .

IV. ARCHITERTURE OF CELLULAR COMMUNICATION FOR USING MMWAVES

5G CELLULAR NETWORK ARCHITECTURE

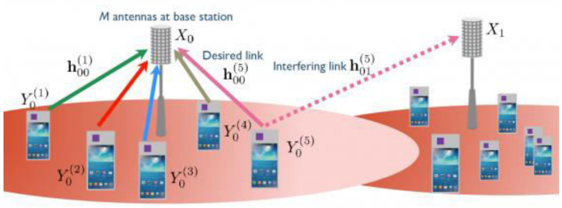
To meet the demands of the user for 5G system, a drastic change in the strategy of designing the 5G wireless cellular architecture is required. This idea will be supported with the help of massive MIMO technology which uses multiple antennas at the transmitter and receiver. They offer many benefits in practicle wireless communications including increase in capacity and spectral effiency,reduction of fading ,improvement resistance to interference.

Since present MIMO systems are using either two or four antennas, but the idea of massive MIMO systems has the idea of utilizing the advantages of large array antenna elements in terms of huge capacity gains.

To build or construct a large massive MIMO network, firstly the outside base stations will be fitted with large antenna arrays and among them some are dispersed around the hexagonal cell and linked to the base station through optical fiber cables, aided with massive MIMO technologies. The mobile users present outside are usually fitted with a certain number of antenna units but with cooperation a large virtual antenna array can be constructed, which together with antenna arrays of base station form virtual massive MIMO links. With a rapid increase in the number of connected devices, some challenges appear which will be responded by increasing capacity and by improving energy effciency, cost and spectrum utilization as well as providing better scalability for handling the increasing number of the connected devices.

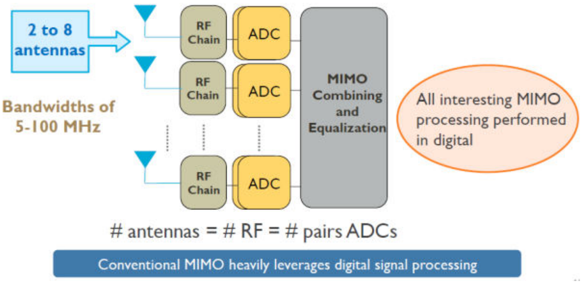
A. MASSIVE MIMO

The Massive MIMO system uses arrays of antenna containing few hundred antennas which are at the same time in one time, frequency slot serving many tens of user terminals. The main objective of Massive MIMO technology is to extract all the benefits of MIMO but on a larger scale. Massive MIMO depends on spatial multiplexing, which further depends on the base station to have channel state information, both on the uplink as well as on the downlink. In case of downlink, it is not easy, but in case of uplink, it is easy, as the terminals send pilots. On the basis of pilots, the channel response of each terminal is estimated. In conventional MIMO systems, the base station sends the pilot waveforms to the terminals and based on these, the terminal estimate the channel, quantize it and feedback them to the base station. This process is not viable for massive MIMO systems, especially in high mobility conditions because of two reasons. Firstly the downlink pilots from the base station must be orthogonal among the antennas, due to which the requirement of time, frequency slots for the downlink pilots increases with the increase in the number. of antennas. So Massive MIMO systems would now require a large number of similar slots as compared to the conventional MIMO system [6]. Secondly, as the number of base station antennas increases the number of the channel estimates also increases for each terminal which in turn needed hundred times more uplink slots to feedback the channel responses to the base station. A general solution to this problem is to work in Time Division Duplexing (TDD) mode and depend on the reciprocity amid the uplink and downlink channels .Figure 7. Analytical model for massive MIMO networks in sub-6 GHz and mmWave

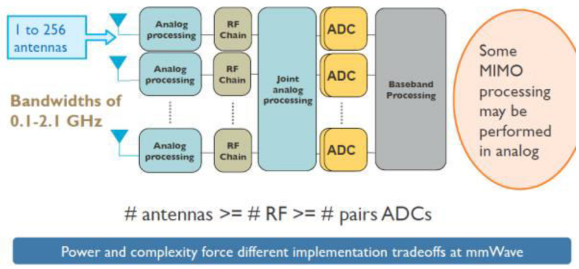


The carrier frequencies for massive MIMO systems, however, are not clear yet; as the propagation channels and hardware constraints will be much different from sub-6 GHz and millimeter wave (mmWave) band.

MIMO receiver at < 6 GHz frequencies



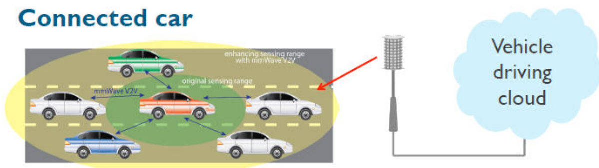
MIMO receiver at mmWave frequencies



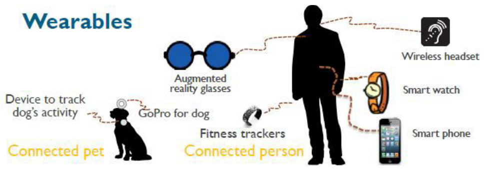
Applications of MIMO communication using MM wavein future wireless communications.

V. APPLICATIONS OF MIMO COMMUNICATION USING MM WAVEIN FUTURE WIRELESS COMMUNICATIONS

- ❖ Single user, multiple user, multi-cell, relay
- ❖ Interference and mobility become more of a challenge



- ❖ Attractive for vehicle-to-vehicle (V2V) and vehicle-to-infrastructure (V2I)
- ❖ Enhanced local sensing capability in connected cars
- ➢ Share high rate sensor data: radar, LIDAR, video, IR video, other sensors
- ➢ Data fusion from other cars can enlarge the sensing range
- ❖ Enables the transition from driver assisted to autonomous vehicles



- ❖ Multiple communicating devices in and around the body
- ➢ 5 or more devices per person based on market trends trend
- ❖ MmWave solves two critical problems
- ➢ Provides high data rates for high-end devices
- ➢ Provides reasonable isolation for low-end devices

VI. CONCLUSION

In this paper, a detailed survey has been done on the performance requirements of future generation 5G wireless cellular communication systems which will mainly use millimeter waves . The need of using mmwaves and the challenges behind using mm waves in 5G wireless cellular communications is explained. The A 5G wireless network architecture has been explained in this paper with massive MIMO Applications of MIMO communication using MM wavein future wireless communications Technology such as single user,multiple user,vehical-to-vehical (V2V) and vehical-to-infrastructure (V2I), enchanced local sensing capability in connected cars etc have been explained. This paper may be giving a good platform to motivate the researchers which promises many research opportunities for mmwaves such as multiuser hybrid precoding, broadband channel models, beam training,models for RF impairment,synchronization etc.

ACKNOWLEDGMENT

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A LOW POWER DBI BASED CRC DESIGN USING GDI TECHNOLOGY

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Abstract—**In this paper, we implemented the data bus inversion cyclic redundancy check using Gate Diffusion input technique. Initially to reduce signaling power in the single-ended interface data bus inversion (DBI) is required, in which the state of the data to be transmitted may or may not be inverted prior to transmission. a new CRC methodology which is based on the DBI is to reduce the CRC calculation delay time and area overhead for high-speed memory devices. GDI logic is introduced as an alternative to CMOS logic. It is a low power design technique which offers the implementation of the logic function with fewer numbers of transistors. GDI gates provide reduced voltage swing at their outputs, In GDI based CRC no of transistors are reduced and power consumption and area is decreased.**

Keywords*—*Data Bus Inversion, gate diffusion input,Cyclick redundency check.

I. INTRODUCTION:

In DBI transmitted data may or may not inverted depends up on encoding algorithm, prior to transmission. The DBI consists of full adder and from this we can extract the information related to the even number of data. To reduce the SSO (Simultaneous Switching Output) noise and improve the signal integrity for the high speed data rate communication system. Cyclic redundancy check code is very useful for detecting and correcting the errors in the signals or data. A low power high speed CRC's plays a vital role in Communication systems [1]. To get computer system process speed , valid data window is required and it is a crucial part for reliable data transmission and also it is the main reason to cause error between semiconductor memory device and computer system. If any device uses low electrical power, its importance increases very rapidly. The semiconductors such as DDR4 SDRAM (Double Data Rate 4 synchronous dynamic Random Access memory) and GDDR5 (Graphic DDR5) required very high speed data rate and low power, CRC(cyclic Redundancy Check) is required[2,3].

Moreover, the difficulty becomes worse in the low power memory device.[2,4],The DBI based CRC reduces the CRC calculation delay time and area overhead for high speed memory devices, and the DBI data can be utilized to detect the data bit errors[5]. This method used for high speed memory devices and quantitatively analyzes the improvements and the errors detection.

The basic GDI cell is shown in Fig:1 though it looks similar to conventional CMOS inverter the source/drain diffusion input of both PMOS and NMOS transistor is different. In conventional inverter circuit, source and drain diffusion input of PMOS and NMOS transistors are always tied at VDD and GND potential, respectively. The diffusion terminal of conventional CMOS inverter acts as an external input in the GDI cell. It helps in the realization of various Boolean functions such as AND, OR, MUX, INVERTER. In this work DBI based CRC is implemented for low power and small area applications.

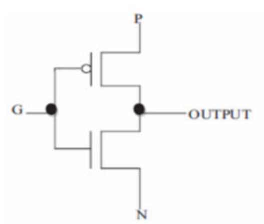

Fig1:Basic GDI Cell

Table1: Different logic function realization using GDI cell

N	P	G	OUT	Function
0	B	A	ÃB	F1
B	1	A	Ã+B	F2
1	B	A	A+B	OR
B	0	A	AB	AND
C	B	A	ÃB+AC	MUX
0	1	A	Ã	NOT

II. FULL SWING XOR GATE

The main drawback GDI cell is low swing, to provide full swing at the output the proposed XOR gate is designed using 4 transistors (excluding the inverter for complementary input signal). The schematic of XOR gate using GDI logic with full swing is shown in Fig.

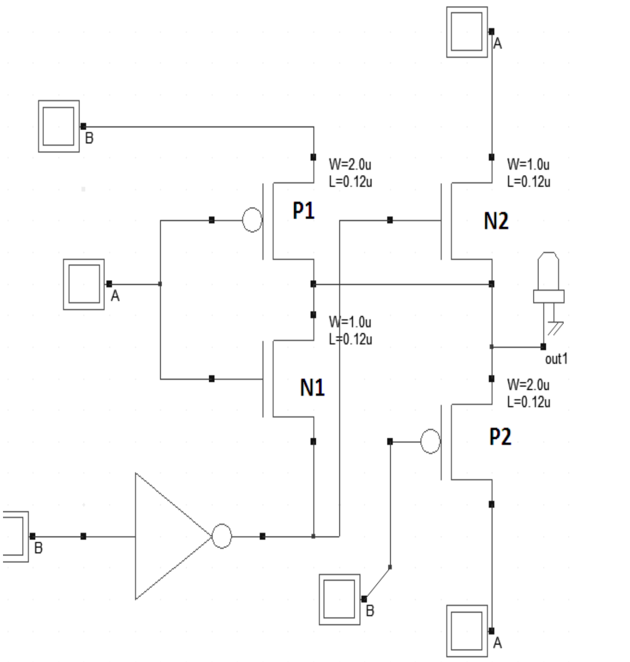

Fig2: Full swing XOR Gate

The operation of proposed XOR gate is explained as follows: The existing design lacks in full swing operation for two cases when AB = 00 and 10. The techniques presented in the literature directly use supply rail VDD for strong ‘1’ and VSS for strong ‘0’. But the proposed design does not use supply rails either GND or VDD for obtaining the perfect output. It uses input, but only with proper biasing of a necessary transistor, which may be either PMOS or NMOS. This in turn would depend on the input level, to mitigate the threshold voltage loss, which occurs in conventional XOR design.

For AB = 00, transistor P1 (PMOS), P3 (NMOS) and P4 (PMOS) conduct. The P3 transistor is responsible for delivering strong ‘0’. Likewise, another case when AB = 10 , transistor P2 (NMOS), P3 (NMOS) and P4 (PMOS) work for the given input, in which P4 passes strong ‘1’ to the output. Whereas in other cases, namely AB = 01 and 11, the transistors P3 and P4 do not change the output potential. Hence, the correct output for XOR gate is attained.

III. PROPOSED GDI BASED DBI- CRC

The DBI consists of full adder and from this we can extract the information related to the even number of data. To reduce the SSO (Simultaneous Switching Output) noise and improve the signal integrity for the high speed data rate communication system, DBI features has been used.

If more than four bits of a byte lane are Low then DBI invert output data. Hence, DBI enables fewer bits switching, which results in less noise and a better data eye. Moreover, we can extract the number of double and sextuple(6 bit) of ‘1’ bits from the DBI feature. And also it can extract the position of double 1’s and sextuple 1’s. This information is applied to the conventional CRC to improve double bits error detection coverage. Conventional DBI based CRC scheme is shown in the figure 3.

When data write to the memory, DB_s signal will send from system to memory and then check the data error between DB_s and DB_m. Simultaneously, calculate the CRC syndrome polynomial S0~S7 in the memory. Finally, the odd data error can be detected by comparing the system data CRC_s and CRC_m. CRC_s is the system data, which is send to the memory and CRC_m is the generated data in the memory which is based on the DQ data in the memory. Also, the partial even data error can be detected by comparing the system data DB_s/DBI_s and DB_m/DBI_m. Figure 2 shows the conventional CRC configuration.

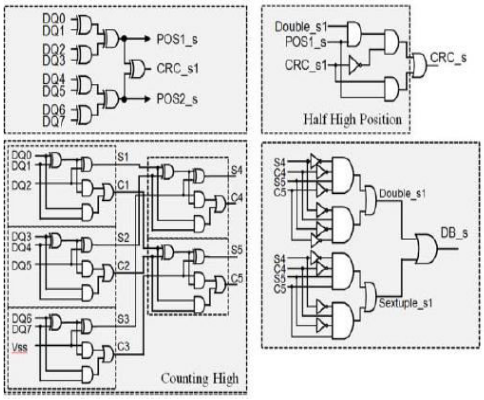

Fig3: Conventional CRC scheme

The expression (1) shows the whole error detection components.

$$\left.\begin{aligned} E1 &= DBI\_s \textbf{ XOR } DBI\_m, \\ E2 &= DB\_s \textbf{ XOR } DB\_m, \quad (1) \\ E3 &= CRC\_s \textbf{ XOR } CRC\_m \end{aligned}\right\}$$

Therefore $Error = E1 + E2 + E3$

Conventional DBI based CRC bit mapping configuration needs to change from the DDR4 data configuration.

IV: PROPOSED GDI BASED DBI-CRC SCHEME

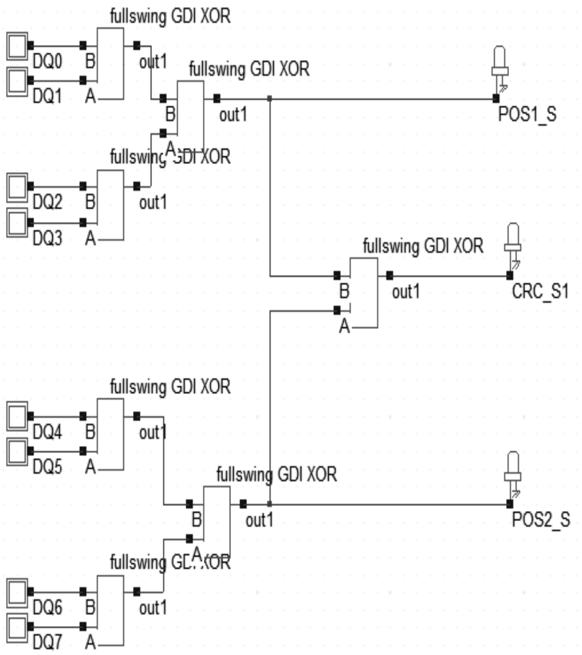

Fig4: GDI CRC Generator

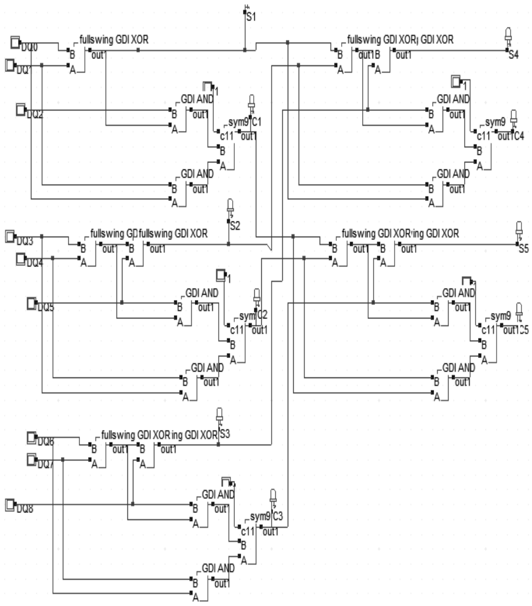

Fig5:GDI CRC count High

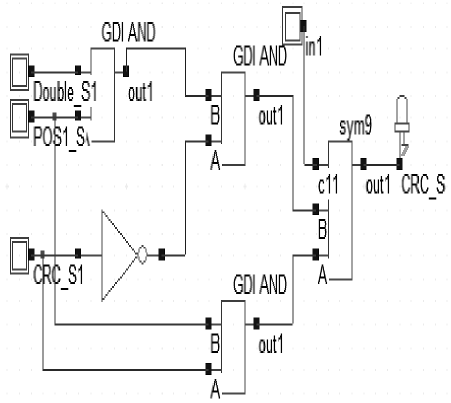

Fig6:Half High Position

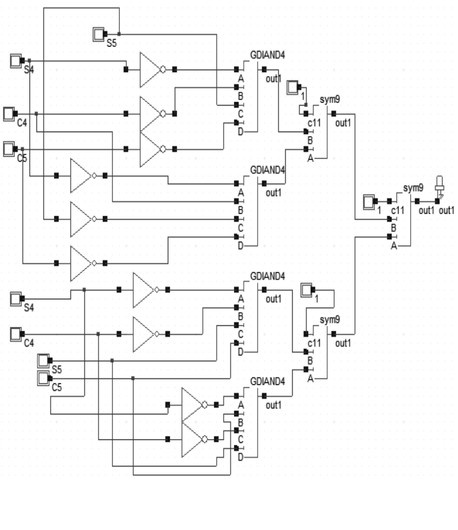

Fig7:Error Detector

V. SIMULATION RESULTS:

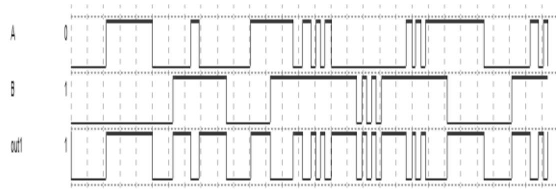

Fig8: Full swing XOR gate using GDI Logic

In an existing design the output swing is poor when inputs AB=00 and 01, but using full swing GDI output swing is HIGH for all four inputs.

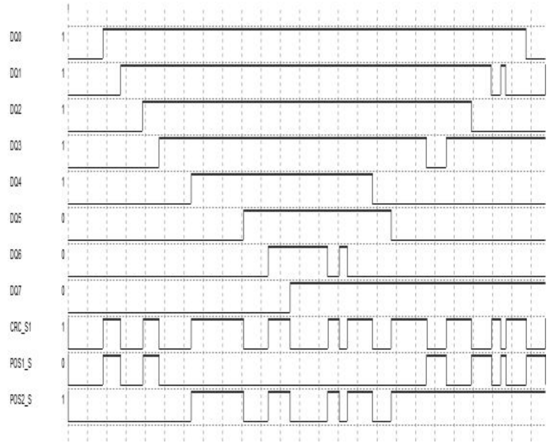

Fig9: CRC Generator

The figure9 shows the CRC generator output. we have taken DQ0- DQ7(8 bit data) as input data and we obtained three outputs namely POS1_S(Position 1), CRC_S1(CRC Code) and POS2_S(Position 2).

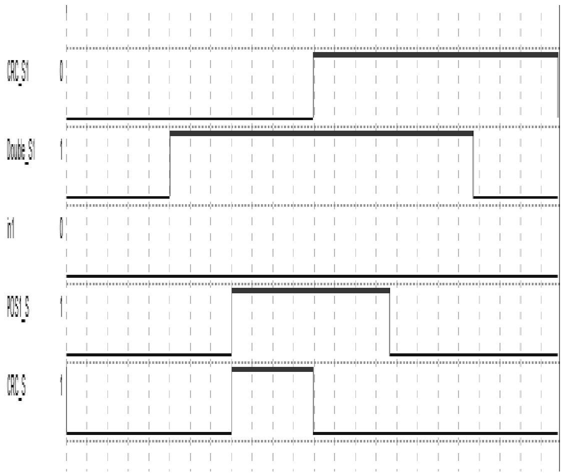

Fig10: Half High Position

The figure10 shows the Half High Position output. We have taken CRC_S1 (CRC Code), POS1_S (Position 1), DOUBLE_S1 (Double error detection) and IN1 as inputs and we observed that CRC_S is fixed.

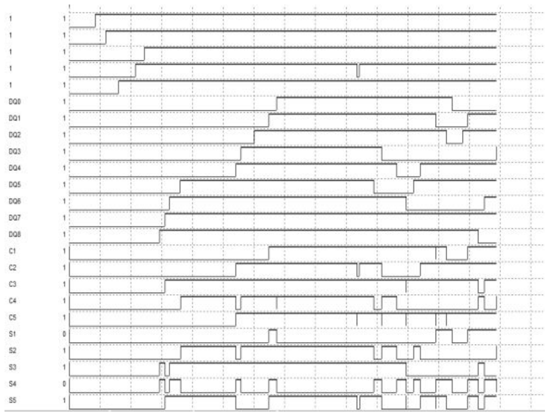

Fig11: Counting High

The figure11 shows the Counting High output. We have taken DQ0- DQ7 (8 bit data) as input data and we obtained outputs namely S1-S5, C1-C5 and HIGH.

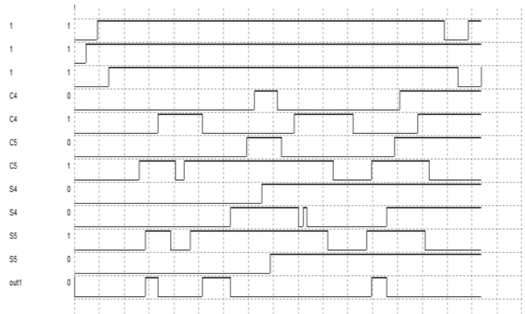

Fig12: Error Detector

The figure12 shows the Error Detector output. We have taken S4, S5, C4, C5 as inputs and we obtained output namely out1(DB_S)

VI. CONCLUSION

. All blocks in the CRC scheme are implemented using GDI technology. The proposed DBI based CRC using GDI technology is taking less no of transistors and area where compare to conventional DBI based CRC.

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A STUDY ON MINORITY COSTUME IMAGE RETRIEVAL BY FUSION OF COLOR HISTOGRAM AND EDGE ORIENTATION HISTOGRAM

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Abstract—**It has very important practical significance to analyze and research minority costume from the perspective of computer vision for minority culture protection and inheritance. As first exploration in minority costume image retrieval, this Project proposed a novel image feature representation method to describe the rich information of minority costume image. Firstly, the color histogram and edge orientation histogram are calculated for divided sub-blocks of minority costume image. Then, the final feature vector for minority costume image is formed by effective fusion of color histogram and edge orientation histogram. Finally, the improved Canberra distance is introduced to measure the similarity between query image and retrieval image. We have evaluated the performances of the proposed algorithm on self-build minority costume image dataset, and the experimental results show that our method can effectively express the integrated feature of minority costume images, including color, texture, shape and spatial information. Compared with some conventional methods, our method has higher and stable retrieval accuracy.**

I. INTRODUCTION

China is a country consisting of 56 ethnic groups, and each of them has its own apparel style with distinct ethnic characteristics, due to the influence of different culture, traditions, and geographical feature. The minority costume is the important symbol of the ethnic group identification and the precious wealth of the Chinese nation. However, with the acceleration of global economic and political integration in China, various minority costume cultural traditions have been rapidly disappearing. This prompted people to think the survival of minority costume under the new historical situation. For now, the minority costumes are mainly protected by museums statically. Compared with the traditional protection mode of physical originals in museums, digital protection has longer protection time and promotes minority costume culture more conveniently. Content-based image retrieval is a very important topic in the field of pattern recognition and artificial intelligence. It has been successfully applied to many fields, such as medical diagnosis, textiles industry and so on. The minority costumes of same nation have their own distinguished characters (unified tone, style and patterns.), which make them more advantageous than ordinary natural images in image processing. Therefore, it is of great importance to analyze the visual features of minority costumes. In this paper, the digital protection of national costume is studied from the perspective of computer vision.

Although national minority clothing image have complex visual features, the main characteristics still are clothing color, fabric texture and totem shape, which are in accordance with the image feature in computer vision. So we can use traditional feature extraction algorithms to extract the features of minority costume images. At present, a large number of approaches on extraction of color, texture and shape features have been put forward and have already obtained good results in many fields.

Color is the most dominant and distinguishing visual feature. The existing color feature extraction methods include color histogram , color moment , color coherence vector and color correlogram .In the current version of the MPEG-7 Final Committee Draft, several color descriptors have been approved including number of histogram descriptors . Texture is used to specify the roughness or coarseness of object surface and described as a pattern with some kind of regularity. Many researchers have put forward various algorithms for texture analysis, such as the famous gray level co-occurrence matrix (GLCM), local binary patterns (LBP) , local directional patterns (LDP) , and so on. With the continuously expanding of the application field, new theory, like the theory of wavelet, is introduced. And in 1996, Tai Sing Lee used Gabor filters to extract texture features. Shape is the most essential feature of the object. The classic shape descriptors are the Hu moment invariants, the Fourier transform coefficients and the histogram of oriented gradients (HOG).

The minority costume image have very complex visual features, which make it more difficult to be expressed by single feature extraction algorithm. So our goal is to design a feature extraction algorithm based on multi-features to express the information of minority image comprehensively. A lot of image feature extraction algorithms based on multi-features have been proposed in recent years. In 2010, Guang-Hai Liu presents a novel image feature representation method, called multi-texton histogram , for image retrieval. It integrates the advantages of co-occurrence matrix and histogram by representing the attribute of co-occurrence matrix using histogram. Micro-structure descriptor proposed by Guang-Hai Liu in 2011 is built based on the underlying colors in micro-structures with similar edge orientation. It effectively integrates color, texture, shape and color layout information as a whole for image retrieval. Guang-Hai Liu also proposed color difference histogram in 2013, which count the perceptually uniform color difference between two points under different backgrounds with regard to colors and edge orientations in L*a*b* color space. The image feature extraction algorithms mentioned above have all achieved high retrieval accuracy in the Corel image database.

In view of many image feature extraction algorithms based on multi-features have been successfully applied in image retrieval, this paper presents a comprehensive feature descriptor to express the rich visual features presented in minority costume image. This descriptor is represented by effective fusion of color histogram and edge orientation histogram. It's implied in experimental results that the image representation techniques used in our method are an effective way of integrating low-level features into a whole.

II. RELATED WORKS

Currently, the research work of minority costume image retrieval is still in its infancy and exploration stage. We are the first ones to conduct exploratory research on the retrieval technology of minority costume image. In this paper, we construct a minority costume image dataset, in which some images are taken by ourselves and some are from the internet. Most of the minority costumes in these images are dressed by minority people or human body model, and some are photographs of tiled minority costumes. Every ethnic group has its own costume style, so we can distinguish between ethnic groups by their costumes. After a series of researches on the characteristics of minority costumes in Yunnan, we choose the six most characteristic ethnic groups' costume as the research object, including Bai nationality, Jingpo nationality, Hani nationality, Miao nationality, Bouyei nationality and Va nationality. For each nationality we collect 100 costume images and preprocess them to size 128×96 or 96×128 in JPEG format. Figure 1 shows some image examples in the minority costume image dataset.

For now, no researcher has conducted exploratory research on the retrieval technology of minority costume image. Nevertheless, many scholars have researched on the image processing technology of ordinary clothing image (especially in e-commerce field) from the perspective of computer vision. Choi Yoo-Joo presents a novel approach to retrieval the person image that contain the identical clothing to a query image from the image set captured by multiple CCTV camera. Firstly, the clothing area is found based on the position of the face area; Then a feature vector is built for the clothing area, which composed by six color histograms of six sub-regions defined in the clothing area. Wang Hai-long presents a method of contour feature extraction, expression and matching to implement clothing image retrieval comprehensively, where the clothing image is from e-commerce websites. Chen jia-lin presents an interactive clothing retrieval system, which supports query by a real-world image with target clothing and returns real-world images with similar clothing.

A novel clothing shape feature is proposed to describe the shape of clothing in human-oriented coordinate system. And a supervised method is also proposed for learning a weighting matrix to minimize the intra-class distance while maximize the inter-class distance. Wang Yatong designs and implements an image querying and retrieval system based on color feature for e-commerce apparel. The paper compares a variety of color feature extraction methods and similarity measure methods. Experiments show that Euclidean metric and global color histogram using RGB space are relatively appropriate for clothing image search. In general, all above algorithms are designed to describe the feature of ordinary clothing image from perspective of color, shape, and texture.

Figure 1. An example of the minority costume image dataset

As an exploratory study, the contribution of this paper includes: (1) We construct a minority costume image dataset of six Yunnan nationalities; (2) We propose a global feature extraction method without any segmentation to express the minority costume image feature; (3) We propose an improved Canberra distance to measure the similarity between two minority costume images; (4) We design and implement a web-based minority costume image retrieval system.

III. FEATURE EXTRACTION OF MINORITY COSTUME IMAGE

A. Calculation of Color Histogram

Color is an important visual attribute for both human perception and computer vision and it is widely used in image retrieval. The color histogram is one of the most direct and the most effective color feature representation . It has advantages of transform invariant, rotate invariant and scale invariant and has been widely used in image retrieval. But it lacks spatial information. This paper incorporates spatial information to it by combining the color histograms for several sub-blocks defined in the minority clothing image. An appropriate color space and quantization must be specified along with the histogram representation. In this paper, three color spaces (RGB, HSV and CIE L*a*b*) with different quantification number are used to test the performance of our Method. The experimental results in Tables 1-3 demonstrate that the RGB color space with 8×4×4=128 quantification number is the best choice in our framework. For an image with a size of $M \times N$, we set the color quantification number to L and denote the image by the equation C□ x, y□($x \in [0, N)$ $y \in [0, M)$). The value range of C□ x, y□ is $[0, L)$. We divide the image to n blocks. The color values of each block is denoted by C_i (x, y),($i \in [0, n)$)

where num_j is the number of pixels in a sub-block whose color value is quantified to j.

B. Calculation of Edge Orientation Histogram

In the system of theory on computer vision, edge detection of image plays an important role. This paper construct a feature descriptor namely edge orientation histogram, which can be seen as a texture feature and also a shape feature. The classic edge detection operator are Sobel, Roberts, Prewitt and Canny. Sobel is one of the most popular operator , which is named after Irwin Sobel and Gary Feldman. The Sobel operator is based on convolving the image with a small, separable, and integer valued filter in the horizontal and vertical directions and is therefore relatively inexpensive in terms of computations. The operator uses two 3×3 kernels which are convolved with the original image to calculate approximations of the derivatives - one for horizontal changes, and one for vertical.

In practical applications, the maximum of the gradient direction is taken. To facilitate implementation, we project it into the interval $[0,2\pi]$. After the edge orientation F (x, y) of each pixel has been computed, the orientations are uniformly quantized into m bins, where m □ 12,18, 24, 30, 36 . Let $H(Q)$ □$[H_1 (Q), H_2 (Q),..., H_M (Q)]$ and $H(T)$ □$[H_1(T), H_2 (T),..., H_M(T)]$ be the feature of query image Q and image T in the database. Then the retrieval results can be returned by computing a similarity measure of feature vector between query image and every image in the dataset.

There are many distance metrics for similarity measure, like Manhattan distance, Euclidean distance, Chi Square distance, Canberra distance etc. The experiment results in Table 6 show that Canberra distance is a better distance metric for our method than others.

IV. LITERATURE REVIEW

GLCM: Grey-Level Co-occurrence Matrix texture measurements have been the workhorse of image texture since they were proposed by Haralick in the 1970s. To many image analysts, they are a button you push in the software that yields a band whose use improves classification - or not. The original works are necessarily condensed and mathematical, making the process difficult to understand for the student or front-line image analyst. This GLCM texture tutorial was developed to help such people, and it has been used extensively world.

LBP: To examine the influence of beliefs about low back pain (LBP) on reduced productivity at work ("presenteeism") caused by LBP. Negative beliefs about LBP are associated with both work absence and reduced work-productivity. Further investigations should examine their potential as a target for educational interventions when considering initiatives to reduce the socioeconomic costs of LBP.

Figure 2 shows the retrieval results for the query “Miao nationality” with the proposed approach. The top-left image is the query image, and the similar images returned include the query image itself. One error result is retrieved in the fourth row and first column.

Figure 2. Retrieval results for query “Bai nationality”

V. CONCLUSION AND FUTURE WORK

In this project, we studied a novel feature extraction approach for minority costume image retrieval which combines color, texture, shape and spatial features of minority costume image effectively. This method has good retrieval performance and strong adaptability. And it’s much more effective than other algorithms reported earlier in the article, such as GLCM, LBP, LDP, Gabor-based feature descriptor, Hu invariant distance, HOG, MTH, MSD and CDH. Because the local feature of minority costume image are obvious, region-based image retrieval for minority costume image dataset will be studied in future work. Maybe, image segmentation will be considered as an assistant to extract the local feature and semantic feature of minority costume image.

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DESIGN AND IMPLEMENTATION OF SELF-CHECKING CARRY-SELECT ADDER BASED ON TWO-RAIL ENCODING

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Abstract—**We first show that self-checking presented in the above paper does not work for carry-select adder with input bits higher than 2. Then, we present a correct design and show that the resulting overhead almost doubles.**

INTRODUCTION: Self-checking carry-select adder (CSeA) with reduced area overhead was presented by Vasudevan *et al.* [1]. The proposed design seemed to be promising for self-checking CSeA, and therefore, the work was referred to for comparison by Alioto *et al.* [2], Belgacem *et al.* [3] and Wang *et al.* [4]. However, we find that the claim for self-checking is only valid for 2-bit CSeA, and the 6-bit CSeA shown in the paper cannot provide self-checking.

In this paper, we argue the failure of their design model and present a correct self-checking CSeA design based on two-rail encoding. We show that the transistor overhead of the correct model is higher than the one claimed by Vasudevan *et al.* [1].

DESIGN PROBLEM*:* The property used by Vasudevan *et al.* [1] for a pair of full adders can be generalized as follows:

The relation between sum bits calculated with identical in- puts is only dependent on the carry-input, and for comple- mented values of carry-input, we will obtain complemented sum bits (keeping other pairs of input bits identical).

For example, if the sum bits S^1_i and S^0_i are generated by using intermediate carry $C^1_i$and C^0_i, respectively, then according to the above-mentioned statements:

If (C^1_i== ~C^1_i)

Then S^1_i is equal to ~S^0_i;

Where, indicates the bit position, while 1 and 0 in the super- script represent the initial value of . If we analyze the CSeA design, then the above statements are only valid for the least sig- nificant sum bit of a particular CSeA block. This is because the first full-adder in every CSeA block is the only one that will get the complemented values of the carry-input. The remaining full adders will depend on the propagated carry, which may or may not be complementary to each other. Therefore, we cannot say whether the generated sum bits, other than the first full adder, will be inverted to each other or not.

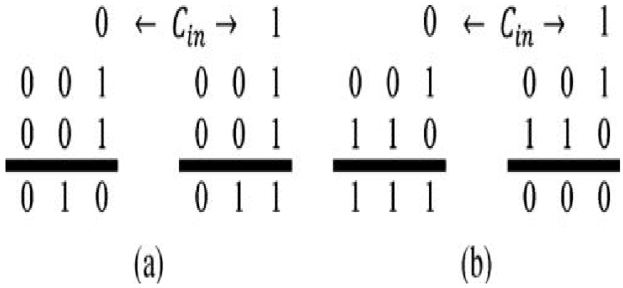


Fig. 1: Examples of 3-bit addition (a) S^0_2=S^1_2 (b) S^0_2=~S^1_2

The possibility of having equal values of propagated carry by the two corresponding adders in a CSeA was neglected by Vasudevan *et al.* Therefore, the approach in [1] fails for CSeA with more than two bits, as shown in Fig. 2[1]. Note that the initial carries, C_1 and C_2 , always complementary to each other because of the design requirement of CSeA, while the inter- mediate carries, C_a and C_b, may or may not be complementary to each other, depending on the conditions of carry propagation. Thus, S_a and S_b will not always be complementary to each other. Therefore, comparing S_a and S_b directly using 2-pair-2-rail-checker (TTRC) will give the wrong indication of faults. Even if there is no fault, the TTRC will indicate a fault. Since the problem in their approach starts from C_a and C_b, we do not discuss the intermediate carries C_x and C_y. Let us consider the binary addition of 3-bit numbers as illustrated in Fig. 1. It can easily be seen from Fig. 1(a) and (b) that the most significant bits may or may not be complementary to each other.

PROPOSED DESIGN SOLUTION: A carry-select adder pre-computes sum bits using two parallel ripple-carry adders (RCAs), with complemented values of the initial C_{in} , and the actual value of the C_{in} will be used to determine the final sum bit. Vasudevan *et al.* utilized both RCAs to obtain the com- plementary behavior of the corresponding sum bits. However, it is possible to perform a logical operation such that one of the RCA blocks should always provide inverted sum bits with respect to the opponent block for checking purposes only. This will provide a more simplified and systematic design, which can be extended easily. In this paper, we will discuss only one possible way in which the sum bits calculated at initial C_{in}=0 are altered, such that they become complementary to the sum bits calculated at an initial C_{in} =1 for comparison.

After close observation, we found that:
Except for the least significant bit, the sum bit computed when initial carry-in equals 0 will be complementary to the corresponding sum bit with an initial carry-in equal to 1 only when all the lower sum bits are equal to logic-1.
In general, we can say that:

If(S^0_1. S^0_2. S^0_3....... $S^0_{(i-1)}$ =1)

Then S^0_i is equal to $\sim S^1_i$;

Else S^0_i is equal to S^1_i;

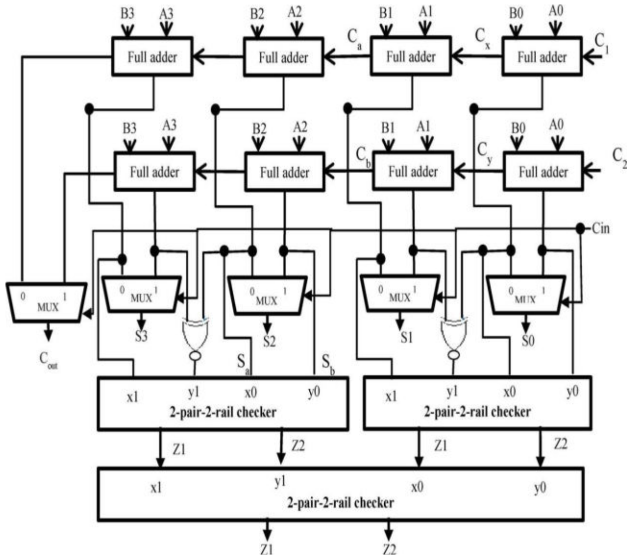


Fig. 2: Faulty design of 4-bit self-testing carry-select adder [1].

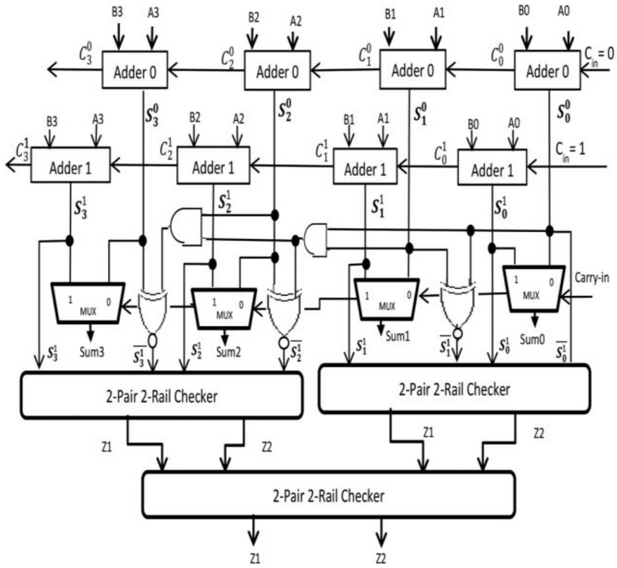


Fig. 3. Corrected design of self-testing carry-select adder with two rail encoding.

Thus, in order to apply TTRC for -bit CSeA, we need to have S^1_n along with its complement, and S^0_n will not always equal to $\sim S^1_n$. If all the(n-1)th sum bits at C_{in}=0 are equal to logic-1, then the value S^0_n is equal to the complement of S^1_n.

In other cases, if any of the (n-1)th sum bits at C_{in}=0 are equal to logic-0, then we take the inverse of S^0_n, so it equals to the complement of S^1_n. Therefore, in (1) we performed an XNOR operation between S^0_n and the product of all lower sum bits computed at the initial C_{in}=0, such that the resultant K will always be equal to ~ S^1_n. The design module shown in Fig. 3 will be used to implement the 4-bit self-checking CSeA.

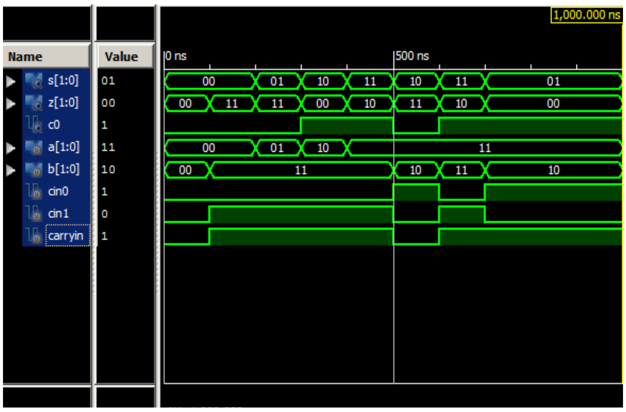
$$K = \overline{S_n^0 \oplus \left(S_{(n-1)}^0 \cdots \cdots S_3^0 \cdot S_2^0 \cdot S_1^0\right)}. \qquad (1)$$

COMPARISON*:* We applied the same technology and im-plementation used by Vasudevan *et al.* [1] for comparison. A standard complementary metal-oxide semiconductor-based AND gate with 6 transistors was used for area computation and the transistor count for full-adder, multiplexer (MUX), XNOR gate and TTRC was taken from Vasudevan *et al.* [1], as given below:

- Full adder – 28 transistors;
- MUX – 12 transistors;
- XNOR – 10 transistors;
- TTRC – 8 transistors.

Number of bits	CSeA without self-checking	Vasudevan et al. faulty design [1]			Corrected self-checking CSeA		
	Transistor required [1]	Transistor required	Transistor overhead	% Transistor overhead	Transistor required	Transistor overhead	% Transistor overhead
4-bit	284	328	44	15.49%	350	66	23.2%
6-bit	420	490	70	16.66%	534	114	27.14%
8-bit	556	652	96	17.26%	718	162	29.14%
16-bit	1100	1300	200	18.18%	1454	354	32.18%
32-bit	2188	2596	408	18.65%	2926	738	33.73%
64-bit	4364	5188	824	18.88%	5870	1506	34.5%

TABLE 1: COMPARISON OF CSEA WITH SELF CHECKING CSEA BEFORE AND AFTER CORRECTION OF VASUDEVAN *ET AL.* [1].



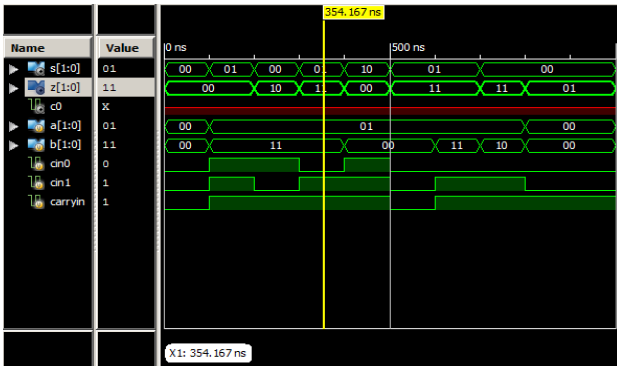


Fig4: Top Module Simulation Results

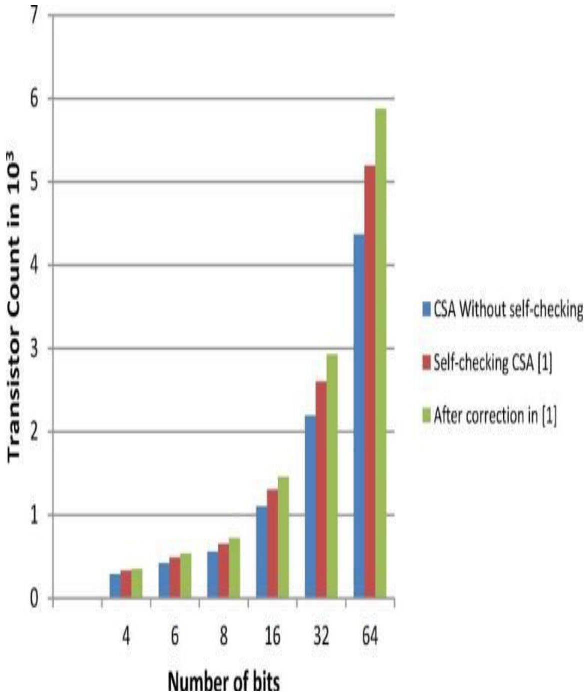


Fig. 5: Comparison with CSeA, self-checking CSeA [1] and our proposed so-lution.

For an n -bit self-checking CSeA, we required (n-2) number of AND gates, (n-1)number of XNOR gates, (n+1)number of MUX, (n-1)number of full adders and number of TTRC, respectively. From Table I, we can see that the difference in tran-sistor overhead for 4- to 64-bit self-checking CSeAs varies from 22 to 682, compared to the faulty self-checking CSeA design by Vasudevan *et al.* [1]. Moreover, the transistor overhead of the corrected self-checking CSeA, as compared to CSeA without self-checking, was found to be 23.2% to 34.5%, whereas in the faulty approach presented by Vasudevan *et al.* [1], the overhead was 15.49% to 18.84%.

A graphical representation for area comparison is shown in Fig. 4. We can see that after correcting the self-checking CSeA design [1], the percent change in transistor count shows an in- creasing trend with the increase in number of bits in the adder.

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MULTICAST ROUTING IN MANETS – A REVIEW

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Abstract—**The conceptual shift in the expectations of the wireless users from voice towards multimedia, from availability towards acceptable quality, and from stand-alone towards group-oriented computing has a significant impact on today's networks in terms of the need for mobility, quality of service (QoS) and multicasting. Mobile Ad hoc Network (MANET) is a collection of self-governing mobile devices which can form a temporary network without any underlying infrastructure using wireless channels. The distinctive characteristics of such networks include mobility of devices resulting in arbitrary network topologies, device/link failures, limited network and device's resources. Designing of routing protocols itself is a difficult task for MANETs due to their dynamic characteristics. Now, incorporating multicasting into MANETs make the design of routing protocols further complex. In this paper, characteristics of MANETs, different multicast routing issues and challenges in MANETs, and finally the performance criteria for evaluating multicast routing protocols for MANETs are discussed.**

Keywords—MANET; routing; multicasting; multicast routing; multicast routing protocol;

I. INTRODUCTION

Ad hoc networking is not a new concept. As a technology for dynamic wireless networks, it has been deployed in military since 1970s. Commercial interest in such networks has recently grown due to the advances in wireless communications. Mobile Ad hoc networks (MANETs) are autonomous communication groups formed by wireless mobile hosts without any established infrastructure or centralized control [1] [4]. They are considered for many commercial applications, including short-term communication for disaster relief, public events, and temporary offices. In MANET communication between neighboring nodes is done directly while the remote nodes are based on multihop wireless links. In Fig. 1., Node A communicates with node C over a multi-hop path, where they must enlist the aid of node B to relay packets between them in order to communicate. The arge circles denote each node's transmission range. In this case, A's circle does not cover C. Each mobile node in the network can act as a sender, receiver or a forwarder of the data. With

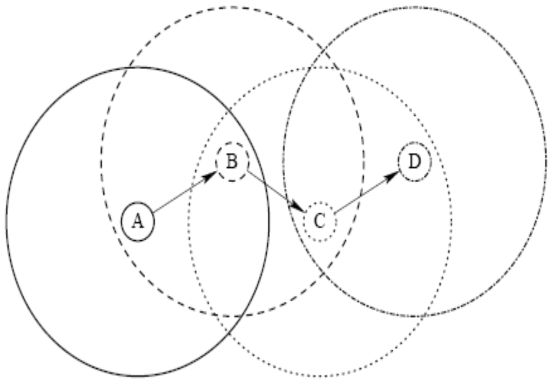

Fig. 1. Multi-hop path in MANET [3]

the use of routing protocol, nodes are capable for communicate with other nodes in the dynamic environment of MANET. Routing in MANET is challenging in the absence of central coordinator as compared to other wireless networks where base station or fixed routers manage routing decisions. Furthermore, multicasting is a promising technique to provide a subset of network nodes with the service they demand while not jeopardizing the bandwidth requirements of others [2]. The advantage of multicasting is that packets are only multiplexed when it is necessary to reach two or more receivers on disjoint paths. As a result of their broadcasting capability, the nodes of an ad hoc network are inherently ready for multicasting.

The paper is structured as follows. In section II, the concepts of MANETs are discussed. Multicasting in MANETs, issues and challenges relaed to it are presented in section III. Section IV providees the performance evaluation metrics of multicast routing protocols. The concluding remarks are given in section V.

II. MANETS

A MANET consists of mobile platforms (e.g., a router with multiple hosts and wireless communication devices), which are simply referred to as "nodes", and are free to move about arbitrarily. The nodes may be located in or on airplanes, ships, trucks, cars, perhaps even on people or very small devices, and there may be multiple hosts per router. Typically, MANET has the following features [3]:

1) Autonomous terminal. In MANET, each mobile terminal is an autonomous node, which may function as both a host and a router. In other words, besides the basic processing ability as a host, the mobile nodes can also perform switching functions as a router. So usually endpoints and switches are indistinguishable in MANET.

2) Distributed operation. Since there is no background network for the central control of the network operations, the control and management of the network is distributed among the nodes. The nodes involved in a MANET should collaborate amongst themselves and each node acts as a relay as needed, to implement functions like security and routing.

3) Multi-hop routing. Basic types of ad hoc routing algorithms can be single-hop and multi-hop, based on different link layer attributes and routing protocols. Single-hop MANET is simpler than multi-hop in terms of structure and implementation, with the cost of lesser functionality and applicability. In single-hop routing, nodes can communicate directly with each other when they are within transmission range of each other; while in multi-hop routing, ad hoc networks must support communication between nodes that are only indirectly connected by a series of wireless hops through other nodes. When delivering data packets from a source to its destination out of the direct wireless transmission range, the packets should be forwarded via one or more intermediate nodes.

4) Dynamic network topology. Since the nodes are mobile, the network topology may change rapidly and unpredictably, and the connectivity among the terminals may vary with time. MANET should adapt to the traffic and propagation conditions as well as the mobility patterns of the mobile network nodes. The mobile nodes in the network dynamically establish routing among themselves as they move around, forming their own network on the fly. Moreover, a user in the MANET may not only operate within the ad hoc network, but may require access to a public fixed network.

5) Fluctuating link capacity. Wireless links will continue to have significantly lower capacity than their wired counterparts. In addition, after accounting for the effects of multiple access, fading, noise, and interference conditions, etc, the realized throughput of wireless communications is often much less than a radio's maximum transmission rate. The nature of high bit-error rates of wireless connection might be more profound in a MANET. One end-to-end path can be shared by several sessions. In some scenarios, the path between any pair of users can traverse multiple wireless links and the links themselves can be heterogeneous.

6) Light-weight terminals. In most cases, the MANET nodes are mobile devices with less CPU processing capability, small memory size, and low power storage. Such devices need optimized algorithms and mechanisms that implement the computing and communicating functions.

Also, the traffic types in ad hoc networks are quite different from those in an infrastructure wireless network, including:

1) Peer-to-peer traffic. Communication between two nodes is within one hop. Network traffic is usually consistent. Example of this kind of scenario is two-sided conferencing application. Fig. 2 shows the example of peer-to-peer communication in MANETs.

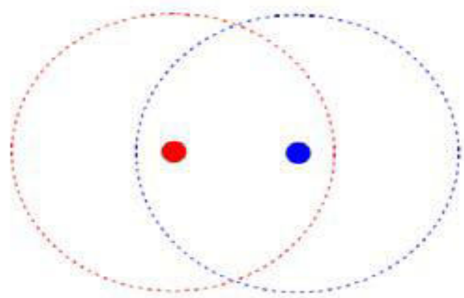

Fig. 2. Example of peer-to-peer communication in MANETs [3]

2) Remote-to-remote traffic. Communication between two nodes is beyond a single hop but maintains a stable route between them. This may be the result of several nodes staying within communication range of each other in a single area or possible moving as a group. Fig. 3 shows the example of remote-to-remote communication in MANETs.

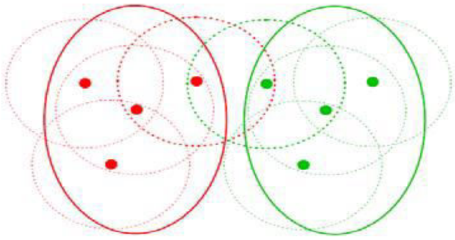

Fig. 3. Example of remote-to-remote communication in MANETs [3]

3) Dynamic Traffic. This occurs when nodes are moving around. Routes must be reconstructed. This results in a poor connectivity and network activity in short bursts. Fig. 3 shows the example of mixed communication in MANETs.

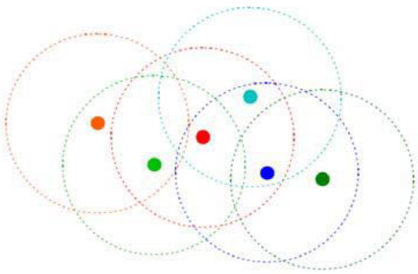

Fig. 4. Example of mixed communication in MANETs [3]

III. MULTICASTING IN MANETS

B. Multicasting

Multicasting is a technique for data routing in networks that allows the same message is forwarded to a group of destinations simultaneously. In mobile ad hoc networks (MANETs), the most challenging issue in multicast routing is to effectively handle the frequent and unpredictable topology changes caused by host mobility, link breakage and host failure. Multicasting is intended for group oriented computing like audio/video conferencing, collaborative works, and etc. Multicasting is an essential technology to efficiently support one-to-many or many-to-many applications. Multicast routing has attracted a lot of attention in the past decade, due to it allows a source to send information to multiple destinations concurrently. Multicasting is the transmission of packets to a group of zero or more hosts called multicast group which is identified by a single destination address. A multicast group is a set of network clients and servers interested in sharing a specific set of data. A typical example of multicast groups is a commander and his soldiers in a battlefield. There are other examples in which multicast groups need to be established. Typically, the membership of a host group is dynamic: that is, the hosts may join and leave groups at any time. There is no restriction on the location or number of members in a host group. A host may be a member of more than one group at a time. A host does not have to be a member of a group to send packets to it. A multicast protocol has the objective of connecting members of the multicast group in an optimal way, by reducing the amount of bandwidth necessary but also considering other issues such as communication delays and reliability.

Multicast routing plays a critical role in most of the new applications such as web-base learning, video conference, and interactive multimedia games. Multicast routing in mobile ad hoc networks poses several challenges due to inherent characteristics of the network such as node mobility, reliability, and scarce resources. The main difficulty in designing a routing protocol for mobile ad hoc networks is the dynamically changing topology, due to the random movement of mobile nodes. The multicast routing protocols designed for wireless mobile ad hoc networks are fundamentally different from those for conventional infrastructure based networks in that these are self-configuring and formed directly by a set of mobile nodes without relying on any established infrastructure. In such networks, the heterogeneity of the hosts makes it difficult to achieve bandwidth efficiency and service flexibility.

In ad hoc networks, there exist several methods upon which classification of the multicast routing protocols is based. Classification based on the underlying structure of the multicast routes, based on the route acquisition time, based on multicast route initiation (based on the responsibility for route construction), and based on the forwarding state maintenance schemes are the well-known representative methods. Underlying multicast route structure is the most popular approach based on which the multicast routing protocols are classified. This classification considers the connectivity of the forwarding paths through which the multicast receivers are connected. This survey presents a new comprehensive classification of the multicast routing protocols based on the underlying route structure. In this classification, the multicasting protocols are categorized as tree-based multicast routing protocols, mesh-based multicasting protocols, hybrid protocols, and stateless protocols. This survey studies the most recent effective protocols of each category with the emphasis on the objectives, performances, costs, advantages, and drawbacks.

C. Design issues and challenges in Multicast routing

The particular features of MANETs make the design of a multicast routing protocol a challenging task. These protocols must deal with a number of issues, including, but not limited to, high dynamic topology, limited and variable capacity, limited energy resources, a high bit error rate, a multihop topology, and the hidden terminal problem. The requirements of existing and future multicast routing protocols and the issues associated with these protocols that should be taken into consideration are listed in what follows:

(i) Topology, Mobility, and Robustness. In MANETs, nodes are free to move anywhere, any time, and at different speeds. The random and continued movement of the nodes leads to a highly dynamic topology, especially in a high-mobility environment. A multicast routing protocol should be robust enough to react quickly with the mobility of the nodes and should adapt to topological changes in order to avoid dropping a data packet during the multicast session, which would create a low packet delivery ratio. It is very important to minimize control overhead while creating and maintaining the multicast group topology, especially in an environment with limited capacity.

(ii) Capacity and Efficiency. Unlike wired networks, MANETs are characterized by scant capacity caused by the noise and interference inherent in wireless transmission and multipath fading. Efficient multicast routing protocols are expected to provide a fair number of control packets transmitted through the network relative to the number of data packets reaching their destination intact, and methods to improve and increase the available capacity need to be considered.

(iii) Energy Consumption. Energy efficiency is an important consideration in such an environment. Nodes in MANETs rely on limited battery power for their energy. Energysaving techniques aimed at minimizing the total power consumption of all nodes in the multicast group (minimize the number of nodes used to establishmulticast connectivity, minimize the number of overhead controls, etc.) and at maximizing the multicast life span should be considered.

(iv) Quality of Service and Resource Management. Providing quality of service (QoS) assurance is one of the greatest challenges in designing algorithms for MANET multicasts. Multicast routing protocols should be able to reserve different network resources to achieve QoS requirements such as, capacity, delay, delay jitter, and packet loss. It is very difficult to meet all QoS requirements at the same time because of the peculiarities of ad hoc networks. Even if this is done, the protocol will be very complex (many routing tables, high control overhead, high energy consumption, etc.). As a result, doing so will not be suitable for these networks with their scarce resources, and resource management and adaptive QoS methods are more convenient than reservation methods for MANETs.

(v) Security and Reliability. Security provisioning is a crucial issue in MANET multicasting due to the broadcast nature of this type of network, the existence of a wireless medium, and the lack of any centralized infrastructure. This makes MANETs vulnerable to eavesdropping, interference, spoofing, and so forth.Multicast routing protocols should take this into account, especially in some applications such as military (battlefield) operations, national crises, and emergency operations. Reliability is particularly important in multicasting, especially in these applications, and it becomesmore difficult to deliver reliable data to group members whose topology varies. A reliable multicasting design depends on the answers of the following three questions. By whom are the errors detected? How are error messages signaled? How are missing packets retransmitted?

(vi) Scalability. A multicast routing protocol should be able to provide an acceptable level of service in a network with a large number of nodes. It is very important to take into account the nondeterministic characteristics (power and capacity limitations, random mobility, etc.) of the MANET environment in coping with this issue.

D. Multicast session life cycle

The various issues involved in a typical multicast session can be identified in the life cycle of the session. During that period, important events can occur: joining/leaving and rejoining a session, and session maintenance. These events can substantially affect the performance of multicast communication. Existing multicast protocols deploy different strategies to handle these events in order to maintain the quality of a multicast session (high packet delivery ratio, minimum end-to-end delay, etc.). This section describes how the session is established and terminated.

Before a source node sends multicast data, it checks whether or not the desired multicast group has been constructed. If it has, it sends multicast data immediately; otherwise, the source node must first construct it. Fig. 5 describes a general method for initializing, constructing, maintaining, and terminating a multicast session.

When a source node has data to send, but no information on a route to a receiver is known, it floods a *Join Request* packet, as shown in Fig. 5. Any node that receives a nonduplicate *Join Request* packet rebroadcasts the *Join Request* packet and stores the last hop node information in its routing table (i.e., a backward route). This process is continued until the *Join Request* packet reaches the receiver. The receiver replies with a *Join Reply* packet. When a node receives a *Join Reply* packet, it checks whether or not the next node address of the *Join Reply* entrymatches its own address. If it matches, the node realizes that it is on the path to the source. Then, it marks itself as a *Forwarder Node* (node *J*, *K*, *X*, and *Y*). The *Join Reply* is propagated until it reaches the source node. This procedure constructs routes from the source node to all receivers. After these processes have been performed, the source can transmit multicast packets to receivers via selected routes and forwarder nodes. This method is known as *source-initiated.*

In a *receiver-initiated* method, if a node wants to join a multicast group (see the receiver at the bottom left of Fig. 5), it broadcasts a *Join Request* packet. If the packet is received by a forwarder node, it replies with a *Join Reply* packet. If the *Join Request* packet is received by an intermediate node (nodes not on the tree, *A*, *B–E*, and *F*), it rebroadcasts the *Join Request* packet. This process is continued until it reaches a node on the tree (forwarder node or member node). The forwarder/member node replies with a *Join Reply* packet.When a node receives a *Join Reply* packet, it checks whether or not the next node address of the *Join Reply* entry matches its own address. If it does, the node realizes that it is on the path to the receiver. It then marks itself as a *Forwarder Node*. The *Join Reply* is propagated until it reaches the receiver node.

There are different mechanisms for maintaining the connectivity of the multicast group. First, the source (core node, group leader) of the multicast group periodically floods a control packet through the network during the *refresh period*; this is called the *Soft-State* method. The control packet is propagated by forwarder nodes, and it eventually reaches all the receivers of the multicast group. Any receiver who wants to leave the multicast group simply does not respond to the control packet; otherwise, it transmits a *Join Reply*. Second, the receiver node periodically floods a control packet through the network. Only source node or forwarder nodes are allowed to respond to the control packet; this is also known as a *Soft-State* method. Third, when a link break is detected between two nodes, a route repair procedure is carried out. One of these two nodes is responsible for detecting and repairing the broken link. This can be done in two ways. In the first, the downstream node (the furthest from the source/core/group leader node) sends a *Join Request* packet to search for its upstream node (receiver on the right-hand side of Fig. 5) by limited flooding. If any node of the desired multicast group (forwarder node or group member) receives the *Join Request* packet, it replies with a *Join Acknowledgment* packet. Otherwise, the *Join Request* packet is rebroadcast by an intermediate node until it reaches a node of the desired multicast group. In the second, the upstream node (the nearest node fromthe source/core/group leader node) initiates a tree construction process (node *X* in Fig. 5). The third mechanism for repairing a broken link is known as a *Hard-State* approach.

The multicast session is terminated by the source/core/group leader node by sending an *End Session* packet, or simply by stopping the transmission of multicast data. If a receiver node wants to leave a multicast group; it sends a *Leave* message or it does not respond to the *Join Request* message sent by the source during the *refresh period.*

IV. PERFORMANCE EVALUATION CRITERIA

There are various criteria for evaluating multicast routing protocols, including, but not limited to, packet delivery ratio (PDR), delivery efficiency, protocol efficiency, average latency, number of total packets transmitted per data received, packet retransmission overhead, data forwarding overhead, and control overhead.

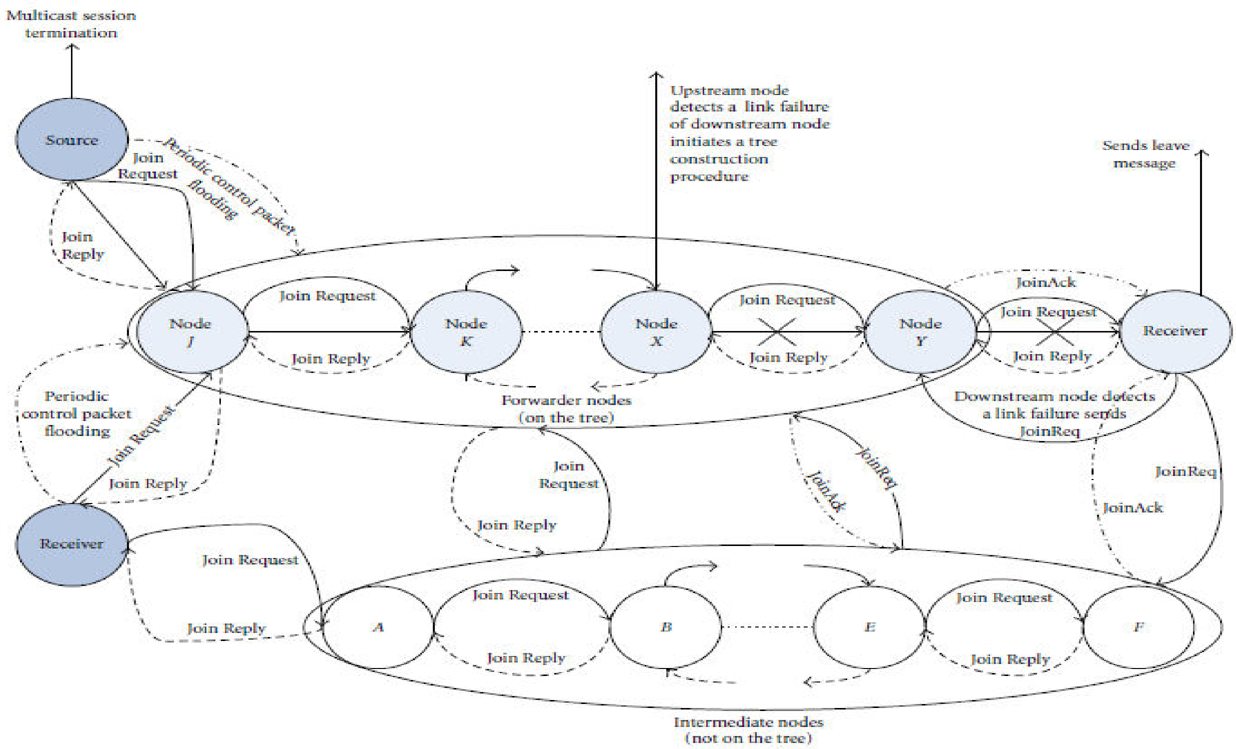

Fig. 5. Multicast Session Life Cycle [3]

(i) *Packet delivery ratio (PDR):* this expresses the number of non duplicate data packets successfully delivered to each destination versus the number of data packets supposed to be received at each destination. It is a metric which can be used as a measure of the effectiveness of the protocol. The higher the PDR, the more efficient and reliable the protocol.

(ii) *Total overhead:* this represents the total number of control packets transmitted and the total number of data packets transmitted versus the total number of data packets delivered. Total overhead is a more important metric than control overhead because we are concerned about the number of packets transmitted to obtain the number of data packets delivered to the receivers, regardless of whether those packets were data or control. It is a measure which shows efficiency in terms of channel access, and is very important in ad hoc networks, since link layer protocols are typically contention-based. (Control overhead represents the number of control pa ckets transmitted (request, reply, acknowledgment) for each data packet that is successfully delivered to the destinations. Control packets are counted at each hop. This metric can be used as a measure of the effectiveness of a multicast protocol. An ineffective multicast protocol will generate a large number of control packets. It shows, relatively, the degree to which an extra wireless channel access is required for the protocol to exchange control information.).

(iii) *Average latency (average end-to-end latency)*: this represents the average time a data packet takes to travel from the transmitter to the receiver. It is a metric which can be used to evaluate the timeliness of the protocol.

(iv) *The data delivery delay between two nodes*: this represents the average time it takes for a data packet to be transmitted from one forwarding node to another.

(v) *Delivery efficiency*: this represents the number of data packets delivered per data packet transmitted. The term of "transmitted" includes "transmitted by sources" as well as "retransmitted by intermediate nodes." The larger its value, the smaller the number of retransmissions.

(vi) *Reachability (RE)*: this represents the number of all destination nodes receiving the data message divided by the total number of all destination nodes that are reachable, directly or indirectly, from the source nodes.

(vii) *Average throughput*: *receiver throughput* is defined as the total amount of data a receiver *R* actually receives from all the senders of themulticast group divided by the time it takes for *R* to receive the last packet. The average over all the receivers is the *average receiver throughput* of the multicast group. The *average throughput* is the average receiver throughput divided by the number of senders.

(viii) *Stress*: the stress of a physical link is the number of identical copies of a multicast packet that needs to traverse the link. This metric quantifies the efficiency of the overlay multicast scheme.

(ix) *Jitter* : defined as the variation in data packet arrival time.

(x) *Packet Drop:*defined as the total number of packets droopped in the network during the transfer of information.

(xi) *Energy Consumption:*defined as the total amount of energy consumed by the nodes in the network.

IV. CONCLUSION

In this paper we have provided a thorough description of MANETs, multicasting in MANETs and the typically used performance criteria for the evaluation of multicast routing protocols of MANETs.

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CONVERGENCE SPEEDS OF THE WEIGHT VECTORS BASED ON ADAPTIVE BEAM FORMING

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Abstract: The paper presents the faster convergence speeds of the weight vectors based on adaptive beam forming. The adaptive beam forming has faster convergence speeds of the weight vectors and much larger output SINRs. The step size of the adaptive algorithm is adjusted between the noise-free *a posteriori* and *a priori* errors to get the faster convergence rate and less misadjustment than the CLMS algorithm. On the other hand, minimizing the square of the augmented noise-free *based on variable step size*, the adaptive algorithm better improvements in the output SINR and accuracy. The results shows, comparison of different adadptive algorithms of MSE and output SINR.

Index Terms—Complex-valued least mean squares (CLMS), convergence speed, shrinkage, steady-state, variable step size, widely linear

I. INTRODUCTION

A smart antenna system combines multiple antenna elements with a signal-processing capability to optimize its radiation pattern automatically in response to the signal environment. In recent years, smart antennas have been considered to be one of the most expected technologies, which are adapted to the demanding high bit-rate or high-quality in broadband commercial wireless communication such as mobile internet or multi-media services [1] [2].In wireless communications, smart antenna systems (or antenna arrays) can be used to suppress multipath fading with antenna diversity and to increase the system capacity by supporting multiple co-channel users in reception and transmission [3]. Adaptive beam forming is the main technique of smart antenna system in which an array of antennas is exploited to achieve maximum reception in a specified direction by estimating the signal arrival from a desired direction (in the presence of noise) while signals of the same frequency from other directions are rejected. Adaptivebeamforming has found numerous applications in radar, sonar, seismology, microphone array speech processing [4]-[8] and in wireless communications. Adaptive beam forming algorithms can be classified into two categories: non-blind adaptive algorithms and blind adaptive algorithms. In this paper, we consider the Non-blind algorithms, Least Mean Square (LMS) and Normalized Least Mean Square (NLMS), and study their performance. Here we propose a new schema called adaptive Normalized Least Mean Square (A-NLMS) to overcome the shortcomings of LMS and NLMS.Simulation shows ANLMS has fastest convergence rate as it can detect the active taps and update the weights of those active elements only. And A-NLMS has narrower beam width with higher gain towards the desired direction compare to other two algorithms.

II. ADAPTIVE BEAMFORMING ALGORITHM

Most of the adaptive beam forming algorithms can be divided into two types according to whether a training signal is used or not : Non-Blind Adaptive algorithm and Blind Adaptive algorithm. In the paper, we consider non blind adaptive algorithm. Blind adaptive algorithm can require training signal, a training signal is sent from the transmitter to the receiver during the training period. Different types of Least-MeanSquare (LMS), Recursive LeastSquare (RLS) algorithms, etc. [16-18] are some examples of non-blind algorithm.

a) *SL-CLMS Algorithm*

The choice of μ in the update of tap weights vector is very critical for SL-CLMS. A small μ will ensure lower steady state MSE, but the algorithm will converge slowly; a large μ can provide faster convergence rate at the cost of higher steady state MSE. Any selection must be a compromise between convergence rate and steady state MSE.

The error signal of SL-CLMS is given by

$$e(k) = s_0(k) - \boldsymbol{w}^H(k)\boldsymbol{x}(k)$$
$$= \epsilon_{\text{opt}}(k) + \boldsymbol{w}_{\text{opt}}^H \boldsymbol{x}(k) - \boldsymbol{w}^H(k)\boldsymbol{x}(k)$$
$$= \epsilon_{\text{opt}}(k) + e_f(k)$$

where

$$e_f(k) = \boldsymbol{w}_{\text{opt}}^H \boldsymbol{x}(k) - \boldsymbol{w}^H(k)\boldsymbol{x}(k) = -\boldsymbol{v}^H(k)\boldsymbol{x}(k)$$

the update weight vk is given by

$$\boldsymbol{v}(k+1) = [\boldsymbol{I}_M - \mu_k \boldsymbol{x}(k)\boldsymbol{x}^H(k)]\boldsymbol{v}(k) + \mu_k \epsilon_{\text{opt}}^*(k)\boldsymbol{x}($$

where xk is the transmit data sequence

B. SWL-CLMS Algorithm

It uses the steepest-descent method and recursively computes and updates the weight vector. Due to the steepest-descend the updated vector will propagate to the vector which causes the least mean square error (MSE) between the beamformer output and the reference signal. The following derivation for the LMS algorithm. The MSE is defined by:

$$\tilde{e}_f(k) = [\boldsymbol{w}_{1,\text{opt}} - \boldsymbol{w}_1(k)]^H \boldsymbol{x}(k)$$
$$+ [\boldsymbol{w}_{2,\text{opt}} - \boldsymbol{w}_2(k)]^H \boldsymbol{x}^*(k)$$
$$= -\boldsymbol{v}_1^H(k)\boldsymbol{x}(k) - \boldsymbol{v}_2^H(k)\boldsymbol{x}^*(k)$$

where

$X^*(t)$ = complex conjugate of the desired signal.

X(t)= received signal from the antenna elements.

w^H=output of the beam form antenna.

$(.)^H$ = Hermetian operator.

The LMS algorithm converges to this optimum Wiener solution. The basic iteration is based on the following simple recursive relation:

W (n+1) =w (n) +μx (n) e (n)

By decreasing μ the precision will improve but it will decrease the adaptation rate. An adaptive μ could solve this issue by starting with a large μ and decrease the factor when the vector converges.

c. . RLS

The convergence speed of the LMS algorithm depends n the Eigen values of array correlation matrix. In an environment yielding an array correlation matrix with large eigen value spread algorithm converges with a slow speed. This problem is solved with the RLS algorithm by replacing the gradient step size μ with a gain matrix $R^{-1}(n)$ at the n^{th} iteration, producing the weight update equation.

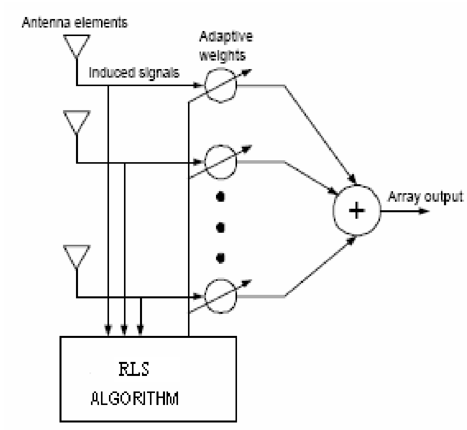

Fig. 1 RLS adaptive beam forming network

W (n) =W (n-1) - R^{-1}(n)
X (n) €*(W (n-1)) Where
R^{-1}(n) is given by

$R^{-1}(n) = \delta_0 \; R^{-1}(n-1)+X(n)*X^H(n)$

Where δ0 denoting a real scalar less than but close to 1, The δ0 is needed for exponential weight of post data and is referred to as the forgetting factor as the update equation leads to de-emphasize the old sample. The quantity 1/(δ0-1) is normally referred to us the algorithm memory, Thus for δ0=0.99 the algorithm memory is close to 100 samples.

The algorithm of RLS algorithm is given by

Step 1) set the initial guess as w (0)

Step 2) Calculate error signal is generated by comparing original unknown signal u (t)

e (t) =x(t) - u (t)

Step 3) Update the weight vector by

W (n) =W (n-1) - R^{-1}(n) X (n) €*(W (n-1))

Where R^{-1}(n) is given by $R^{-1}(n) = \delta_0 \; R^{-1}$(n-

$1)+X(n)*X^H(n)$

Step 4) Go back to <step2> if the procedure is to be continued.

Step 5) Array output $Y(n) = W^h X(n)$

Where $X(n)=S(t)a(\Theta 0)+\sum Ut(t)a(\Theta t)+n(t)$

III. RESULTS

We have implemented the adaptive algorithm by using the MATLAB environment.

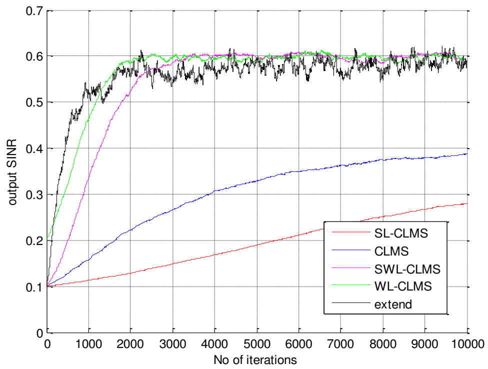

FIG.2. Output SINRs of SL-CLMS, SWL-CLMS, CNLMS, WL-CNLMS, VSS and WL-VSS algorithms

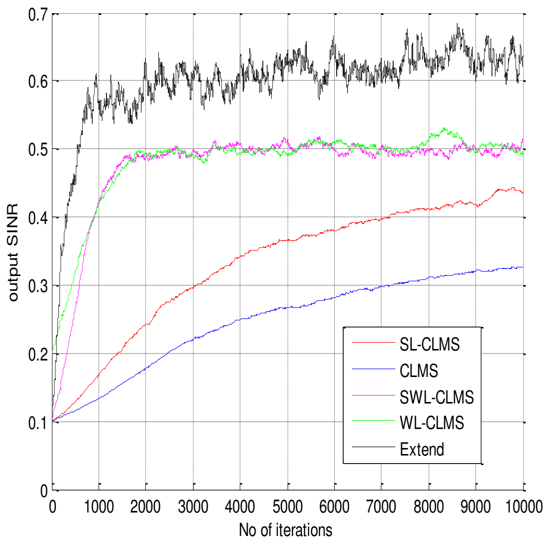

FIG.3. Output SINRs of SL-CLMS, SWL-CLMS, CLMS, WL-CLMS algorithms for different μ .

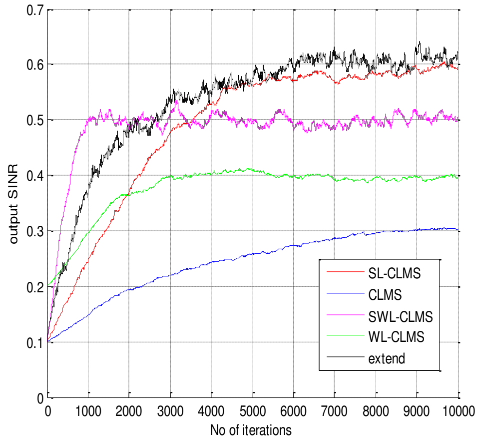

Fig .4. Output SINRs of SL-CLMS, SWL-CLMS, CLMS, WL-CLMS algorithms for different .

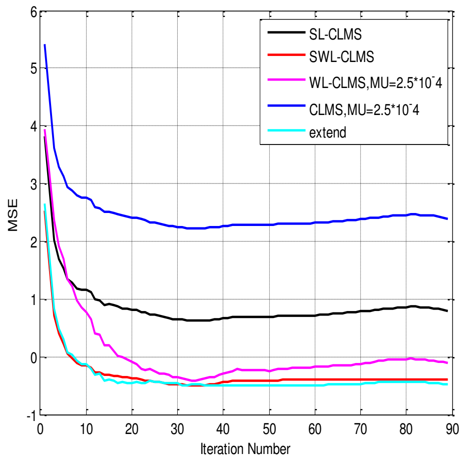

Fig.5. Learning curves: MSEs of SL-CLMS, SWL-CLMS, CLMS and WL-CLMS algorithms Q=1

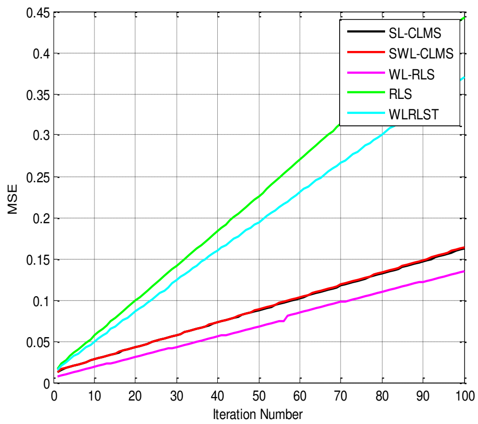

Fig.6. MSE of SL-CLMS, SWL-CLMS, RLS and WL-RLS algorithms versus number of sensors Q=2.

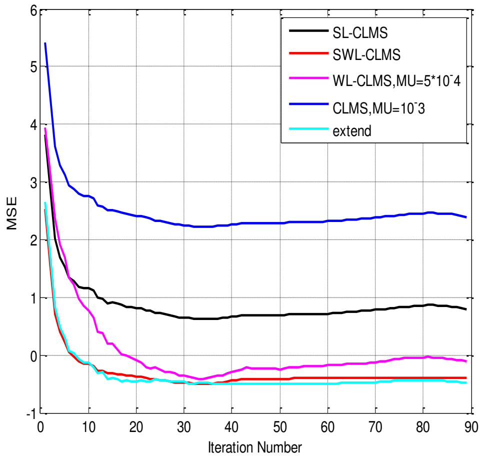

FIG.7. MSE of SL-CLMS, SWL-CLMS, RLS and WL-RLS algorithms versus number of sensors Q=1.

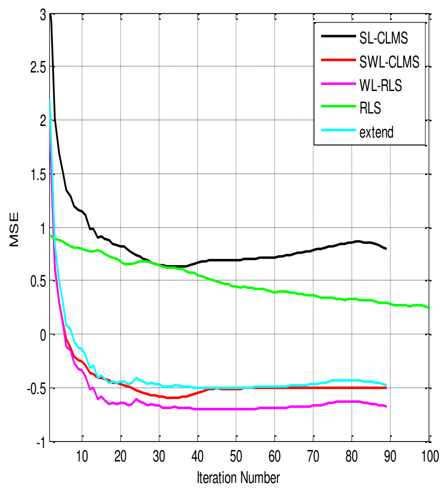

Fig.8. Computational times of SL-CLMS, SWL-CLMS, RLS and WL-RLS algorithms versus number of sensors Q=3.

IV. CONCLUSION

This paper discussed various adaptive beam forming algorithms like SL-CLMS, RLS, SWL-CLMS etc used in smart antennas. The result obtained from the simulations showed that the adaptive algorithm had better convergence compared to LMS, and most efficient algorithm. It can suppress interference and increase the user capacity of a CDMA cellular system. Further through adaptive beam forming, the base station can form narrower beams towards the desired user and nulls towards interfering users, considerably improving the signal-to-interference-plus-noise ratio. Such smart antennas also can be used to achieve different benefits .

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POSTURE RECOGNITION OF STANDING HUMAN BODIES IN STATIC IMAGES

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Abstract— **Recognition of human bodies in images is a challenging task that can facilitate numerous applications, like scene understanding and activity recognition. In order to cope with the highly dimensional pose space, scene complexity, and various human appearances, the majority of existing works require computationally complex training and template matching processes. We propose a bottom-up methodology for automatic extraction of human bodies from single images, in the case of almost upright poses in cluttered environments. The position, dimensions, and color of the face are used for the localization of the human body, construction of the models for the upper and lower body according to anthropometric constraints, and estimation of the skin color. Different levels of segmentation granularity are combined to extract the pose with highest potential. The segments that belong to the human body arise through the joint estimation of the foreground and background during the body part search phases, which alleviates the need for exact shape matching.**

Index Terms — **Adaptive skin detection, Anthropometric constraints (Face detection), Human body segmentation, Multilevel image segmentation (upper and lower body).**

I. INTRODUCTION

Extraction of the human body in unconstrained still images is challenging due to several factors, including shading, image noise, occlusions, background clutter, the high degree of human body deformability, and the unrestricted positions due to in and out of the image plane rotations. Knowledge about the human body region can benefit various tasks, such as determination of the human layout, recognition of actions from static images and sign language recognition. Human body segmentation and silhouette extraction have been a common practice when videos are available in controlled environments, where background information is available, and motion can aid the segmentation through background subtraction. In static images, however, there are no such cues, and the problem of silhouette extraction is much more challenging, especially when we are considering complex cases Moreover, methodologies that are able to work at a frame level can also work for sequences of frames, and facilitate the existing methods for action recognition based on silhouette features and body skeletonization.

The major contributions of this study address upright and not occluded poses.

1) We propose a novel framework for automatic segmentation of human bodies in single images.

2) We combine information gathered from different levels of image segmentation, which allows efficient and robust computations upon groups of pixels that are perceptually correlated.

3Soft anthropometric constraints permeate the whole process and uncover body regions.

4) Without making any assumptions about the fore ground and background, except for the assumptions that sleeves are of similar color to the torso region, and the lower part of the pants is similar to the upper part of the pants, we structure our searching and extraction algorithm based on the premise that colors in body regions appear strongly

II. RELATED WORK

We classify approaches for human body segmentation into the following categories. The first includes *interactive* methods ([10]–[14]) that expect user input in order to discriminate the foreground and background. Interactive segmentation methods are useful for generic applications, and have the potential to produce very accurate results in complex cases. However, since they rely on low-level cues and do not employ object-specific knowledge, they often require user input to guide their process, and are inappropriate for many real-world problems, where automation is necessary. In general, this category differs from the other two, which are automatic and often task specific. The second category includes *top-down* approaches, which are based upon *a priori* knowledge, and use the image content to further refine an initial model. Top-down approaches have been proposed ([15]–[17]) as solutions to the problem of segmenting human bodies from static images. The main characteristic of these approaches is that they require high-level knowledge about the foreground, which in the case of humans is their pose. One method for object recognition and pose estimation is the pictorial structures (PS) model and its variations ([3], [18]–[20]). In general, human body segmentation approaches based on PS models can deal with various poses, but they rely on high-level models that might fail in complex scenarios, restricting the success of the end results. Besides, high-level inference is time consuming and, thus, these methods usually are computationally expensive.

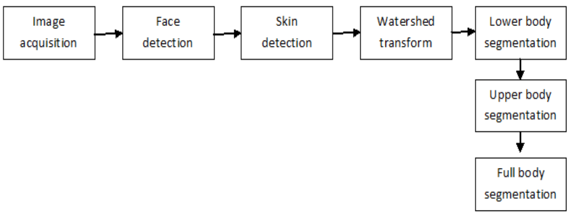

Fig 1:Block diagram of proposed work

III. FACE DETECTION

Localization of the face region in our method is performed using OpenCV's implementation of the Viola–Jones algorithm [33] that achieves both high performance and speed. The algorithm utilizes the Adaboost method on combinations of a vast pool of Haar-like features, which essentially aim in capturing the underlying structure of a human face, regardless of skin color. Since skin probability in our methodology is learned from the face region adaptively, we prefer an algorithm that is based on structural features of the face. The Viola–Jones face detector is prone to false positive detections that can lead to unnecessary activations of our algorithm and faulty skin detections. To refine the results of the algorithm, we propose using the skin detection method presented in [34], and the face detection algorithm presented in [35]. The skin detection method is based on color constancy and a multilayer perceptron neural network trained on images collected under various illumination conditions both indoor and outdoor, and containing skin colors of different ethnic groups. The face detection method is based on facial feature detection and localization using low-level image processing techniques, image segmentation, and graph-based verification of the facial structure.

Fig 2: Input Images i, ii & iii

Fig 3: Face Detection of Input Images i, ii, iii

IV. WATERSHED SEGMENTATON

Watershed Transformation The concept of Watersheds is well known in topography. It was first proposed as a potential method for image segmentation. It is a morphological based method of image segmentation. The gradient magnitude of an image is considered as a topographic surface for the watershed transformation. Watershed lines can be found by different ways. The complete division of the image through watershed transformation relies mostly on a good estimation of image gradients. The result of the watershed transform is de- graded by the background noise and produces the over-segmentation. Also, under segmentation is produced by low-contrast edges generate small magnitude gradients, causing distinct regions to be erroneously merged. There are different ways to find watershed lines. Different approaches may be employed to use the watershed principle for segmentation. Local minima of the gradient of the image may be chosen as markers, in this case an over-segmentation is produced and a second step involves region merging. Marker based watershed transformation make use of specific marker positions which have been either explicitly defined by the user or determined automatically with morphological operators or Watershed segmentation is a common technique for image segmentation.

Fig 4: Multi-level segmentation of images i, ii & iii

V. SKIN DETECTION

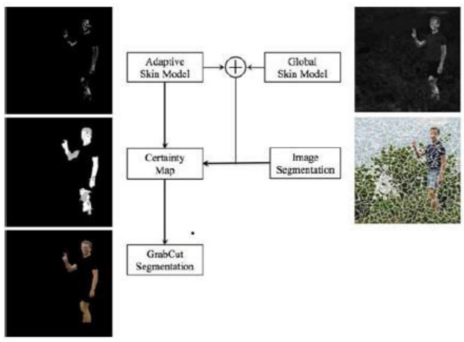

Fig. 3. Skin detection algorithm.

Fig 5: Skin detection algorithm

Among the most prominent obstacles to detecting skin regions in images and video are the skin tone variations due to illumination and ethnicity, skin-like regions and the fact that limbs often do not contain enough contextual information to discriminate them easily. In this study, we propose combining the global detection technique [39] with an appearance model created for each face, to better adapt to the corresponding human's skin color (Fig. 3). The appearance model provides strong discrimination between skin and skin-like pixels, and segmentation cues are used to create regions of uncertainty. Regions of certainty and uncertainty comprise a map that guides the Grab-Cut algorithm, which in turn outputs the final skin regions. False positives are eliminated using anthropometric constraints and body connectivity.

VI. UPPER BODY SEGMENTATION

Fig 6: upper body segmentation of images i, ii & iii

In this section, we present a methodology for extraction of the whole upper human body in single images, extending [40], which dealt with the case, where the torso is almost upright and facing the camera. The only training needed is for the initial step of the process, namely the face detection and a small training set for the global skin detection process. The rest of the methodology is mostly appearance based and relies on the assumption that there is a connection between the human body parts. Processing using super-pixels instead of single pixels, which are acquired by an image segmentation algorithm, yield more accurate results and allow more efficient computations. The initial and most crucial step in our methodology is the detection of the face region, which guides the rest of the process. The information extracted in this step is significant. First, the color of the skin in a person's face can be used to match the rest of his or her visible skin areas, making the skin detection process adaptive to each person. Second, the location of the face provides a strong cue about the rough location of the torso. Here, we deal with cases, where the torso is below the face region, but without strong assumptions about in and out of plane rotations. Third, the size of the face region can further lead to the estimation of the size of body parts according to anthropometric constraints. Face detection here is primarily conducted using the Viola–Jones face detection algorithm for both frontal and side views. Since face detection is the cornerstone of our methodology, we refine the results of the aforementioned method using the face detection algorithm presented in [35].

VII. LOWER BODY EXTRACTION

The algorithm for estimating the lower body part, in order to achieve full body segmentation is very similar to the one for upper body extraction. The difference is the anchor points that initiate the leg searching process. In the case of upper body segmentation, it was the position of the face that aided the estimation of the upper body location. In the case of lower body segmentation, it is the upper body that aids the estimation of the lower body's position. More specifically, the general criterion we employ is that the upper parts of the legs should be underneath and near the torso region. Although the previously estimated UBR provides a solid starting point for the leg localization, different types of clothing like long coats, dresses, or color similarities between the clothes of the upper and lower body might make the torso region appear different (usually longer) than it should be. To better estimate the torso region, we perform a more refined torso fitting process, which does not require extensive computations, since the already estimated shape provides a very good guide.

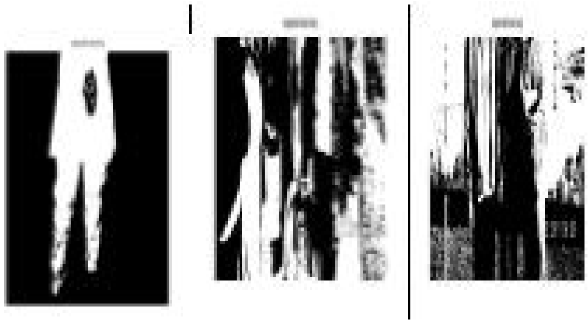

Fig 7: Lower body Segmentation of images i, ii & iii

VIII. RESULTS:

To evaluate our algorithm, we used samples from the publicly available INRIA person dataset [41], which includes people performing everyday activities in outside environments in mostly upright position. This is a challenging dataset, since the photos are taken under various illumination conditions, in heavily cluttered environments and people appear in various types of clothing.

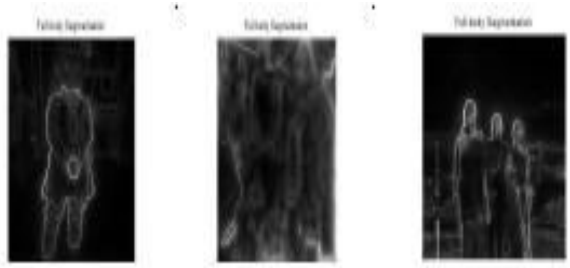

Fig 8: Full body Segmentation of images i, ii & iii

IX. CONCLUSION

The first advantage of our methodology over those tested is that it can automatically localize and segment the human body. Additionally, the final results achieve very good accuracy, even in complex scenarios, and the small standard deviation shows that it is stable. The main advantages of our method are as follows. First, we combine cues from multiple levels of segmentation; therefore, to take into consideration different perceptual groupings from coarse to fine. Second, during our searching process, we try to find arbitrary salient regions that are comprised by segments that appear strongly inside the (hypothesized) foreground rectangles and weakly outside. By considering foreground and background conjunctively, we alleviate the need for exact mask fitting and dense searching, and we allow the masks to be large according to anthropometric constraints so that they may perform sufficient sampling in fewer steps. Third, we demonstrate how soft anthropometric constraints can guide and automate the process in many levels, from efficient mask creation and searching to the refinement of the probabilistic map that leads to the final mask for the body regions. Searching for the upper and lower body parts, as well as the similar process of torso fitting, however, still remain one of the most computationally expensive steps of the methodology.

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ROBUST DT CWT BASED DIBR 3D VIDEO WATERMARKING USING CHROMINANCE EMBEDDING

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Abstract*—The popularity of 3D video is increasing daily due to the availability of low-cost 3D televisions and high-speed Internet access. However, currently the contents of 3D video can be distributed illegally without any protection. For views generated using a depth-image-based rendering technique, the left and right views can be distributed as 3D content, but also the centre, left or right views can be distributed, individually as 2D content. Protecting these views from unauthorized distribution is becoming a very important problem to address. As digital video watermarking is a possible way of achieving this protection, in this paper, we propose a digital watermarking method for depth-image-based rendered 3D video to protect its left and right views as well as the centre view. In this method, the watermark is embedded in both of the chrominance channels of a YUV representation of the centre view using the dual-tree complex wavelet transform. Then, the left and right views are generated from the watermarked centre view and depth map using a depth-image based rendering technique. Finally, the watermark can be extracted from the centre, left and right views in a blind fashion without using the original unwatermarked centre, left or right views. This watermark is robust to geometric distortions, such as upscaling, rotation and cropping, downscaling to an arbitrary resolution and the most common video distortions, including lossy compression and additive noise. Due to the approximate shift invariance characteristic of the dual-tree complex wavelet trans-form, the technique is robust against distortions in the left and right views generated using depth-image based rendering. The proposed method can also survive baseline distance adjustment and both 2D and 3D camcording.

***Index Terms*—Depth-image-based rendering (DIBR), 3D video watermarking, dual-tree complex wavelet transform (DT CWT), robustness, camcording**

INTRODUCTION

Due to the availability of low-cost 3D televisions, the popularity of 3D video content is increasing day by day. There are two major techniques for 3D video representation: stereo imaging and depth-image-based rendering (DIBR). In the former approach, the left and right views are captured using two cameras placed in the positions of two eyes. On the other hand, the DIBR system consists of the centre view and depth map which are transmitted over the channel rather than the left and right colour views. Virtual left and right views are synthesized from the centre view and depth map at the receiver. This approach has the advantage that a depth map can be compressed more efficiently than colour views.

Many illegal copies of digital video productions for cinema release can be found on the Internet or on the street markets before their theatrical release. As an unauthorized copy of a 2D or 3D video can be modified, mass-produced, up-loaded online and distributed globally, protecting its content from illegal distribution is becoming an important issue to be addressed. Although digital video watermarking, which is the process of hiding digital information in a host video signal, is a solution to this problem, there are some important issues such as imperceptibility, blind detection, security and robustness to attacks, that need to be considered during the design of the watermarking algorithm. The watermark is embedded as additional information in the original video and hence may degrade the visual quality of the host video in some manner. A watermarking algorithm is considered to be imperceptible if human eyes cannot distinguish between the original and watermarked video. Hence, the watermarking algorithm must ensure that the watermark embedded in a host video does not affect the visual quality of the watermarked video. As, in most practical applications, the original host video is not available for watermark detection at the decoder, the watermark should be detectable without reference to the original video content, i.e., the detection should be blind. Assuming that a potential attacker has knowledge of the watermark embedding and extraction algorithms, watermark estimation re-modulation (WER), temporal frame averaging (TFA) and multiple watermark embedding are possible attacks that could be used to remove the watermark. A watermarking algorithm is secure only when an unauthorized person is unable to remove the watermark from the watermarked video after these attacks. Furthermore, both 2D and 3D video contents might be subjected (intentionally or unintentionally) to geometric attacks, such as scaling, rotation, and cropping during camcording from a movie theatre. Although pirated high-definition (HD) 2D content may not be compatible with the portable devices such as smart phones, portable multimedia players (PMPs), personal digital assistants (PDAs), and tablets, a pirate can downscale the resolution of the video to a resolution compatible with these devices and distribute the content worldwide through the Internet. These downscaling in resolution and geometric attacks cause loss of video information as well as watermark information. We cannot also ignore the addition of noise and lossy compression as they are common video distortions. Therefore, because a watermark must still be detectable after such events, robustness against the corresponding modifications to the video signal is a very important consideration when designing a watermarking algorithm.

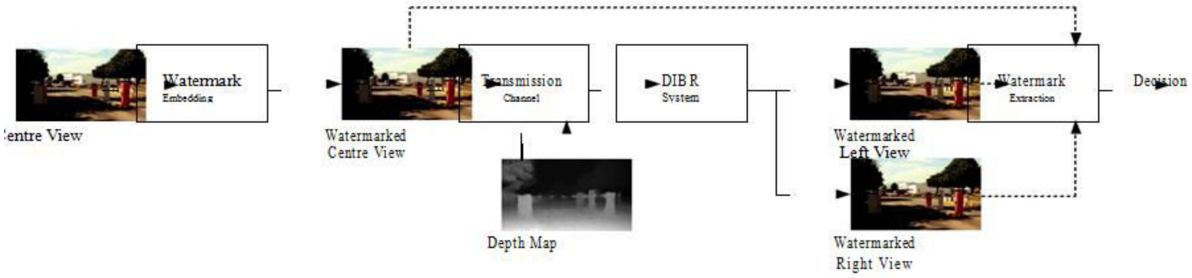

Fig. 1. Diagram of the proposed watermarking system.

In the last few years, a number of image watermarking algorithms have been proposed for DIBR 3D images. As the original image is required to detect the watermark, i.e., it is a non-blind scheme, in a practical application where the original image is not present at the decoder end, this approach is not appropriate. Pei *et al.* proposed a 3D unseen visible watermarking scheme based on a depth no-synthesis-error (D-NOSE) model. This scheme can detect and find which regions of depth images are suitable for watermark embedding. The watermark is embedded into these regions by modifying the depth map with optimal variation pixel values. As the view synthesis is very sensitive to variation of depth values, this scheme mainly focus on synthesis error and the D-NOSE model is used to create a high-quality 3D video for no synthesis error under normal rendering condition. However, this method has a limited defence against commonly used attacks. Three watermarks are embedded in the discrete cosine transform (DCT) domain of the centre view for the left, right and centre views. The virtual left and right views are obtained by rendering the centre view and depth map at the receiver and then the watermarks are extracted from the centre, left and right views. Although this scheme is analyzed for robustness to JPEG compression and additive noise, geometric attacks, which cannot be ignored, are not considered. An algorithm to protect the centre and virtual left and right views, where SIFT-based feature points are exploited to perform synchronization of the watermark, was proposed. This algorithm is robust against some signal processing attacks but cannot survive geometric ones. In another watermarking approach for DIBR 3D images, the watermark is embedded in the level 2 and level 3 coefficients of a 3-level dual-tree complex wavelet transform (DT CWT) decomposition of the luminance (Y) channel of the centre view using a quantization process and then extracted from the centre, left and right views in a blind fashion to protect all views [2]. This approach is robust to scaling and baseline distance adjustment but demonstrates poor performances after rotation and a combination of these attacks. The use of the chrominance channel enhances the imperceptibility of the watermark because, to human eyes, distortion is more noticeable in the luminance channel than the chrominance channels. Therefore, the watermark embedding strength, i.e., robustness to attacks at equal imperceptibility, is higher for the chrominance channel than the luminance channel. In the literature, although some techniques embed the watermark in the chrominance channel, their embedding procedure and applications are different.

In this paper, we propose a blind DIBR 3D video water-marking scheme in which the watermark is embedded in the DT CWT domain of both chrominance (U and V) channels of a YUV representation rather than the Y channel of the centre view. At the decoder, the virtual left and right views are generated from the depth map and watermarked centre view using the DIBR technique and then the watermark is extracted from any one of the three views in a blind fashion. An overall scenario of our proposed watermarking system is depicted in Fig. 1. Some important properties such as perfect reconstruction, shift invariance, and good directional selectivity of the DT CWT, enhance the robustness of the watermark extracted from the virtual left and right views. If a frame is re-sampled after scaling or rotation, the magnitudes of the low-frequency DT CWT coefficients are approximately the same which produces a watermarking algorithm robust against geometric attacks. This imperceptible watermark-ing scheme is resistant to lossy H.264/AVC compression, additive noise, geometric attacks, downscaling to an arbitrary resolution, and camcording of both 2D and 3D videos.

The remainder of this paper is organized as follows: The DIBR system is discussed in Section II. Section III describes the proposed watermarking method. An experimental analysis is given in Section IV and finally Section V concludes this paper.

DEPTH-IMAGE-BASED RENDERING

In a DIBR system, the virtual left and right views can be synthesized using the following three steps: pre-processing of the depth map, image warping, and hole filling. In the pre-processing step, the values of the depth map that range from 0 to 255 are normalized linearly

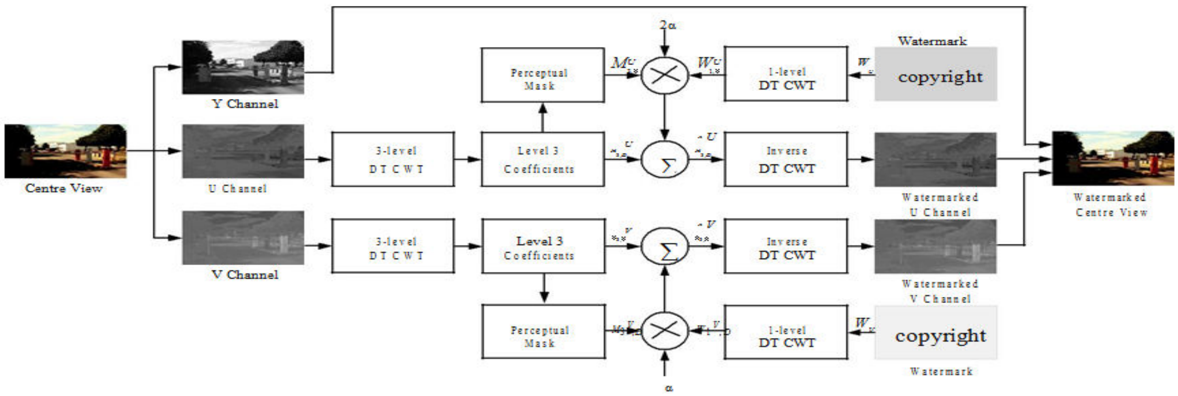

Fig. 2. Watermark embedding in the U and V channels of the centre view.

A Gaussian filter to improve the quality of the synthesized left and right views by minimizing the number of holes, i.e., appropriate smoothing of the depth map provides a better-quality image. In the image warping step, pixel mapping is performed according to the following geometry:

$$x_l = x_c + \frac{t_x}{2} \times \frac{f}{Z},$$

$$x_r = x_c - \frac{t_x}{2} \times \frac{f}{Z} \qquad (1)$$

where x_c, x_l, and x_r are the x-coordinates in the centre, and synthesized left and right views respectively, t_x is the baseline distance, f is the focal

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lengths of the three cameras, and Z is the depth value in the normalized depth map associated with the x-coordinate in the centre image, x_c. Note that, pixel mapping is performed first for the farthest depth value to occlude the farthest parts of the scene by closer objects according to the visibility property of the 3D video.

Hole filling is the final step in the DIBR process and is used to minimize the effects caused by new areas that are exposed in the virtual left and right views after image warping. These areas are usually called disocclusions and background pixel extrapolation is typically used to fill these holes.

PROPOSED WATERMARKING SCHEME

A. Watermark Embedding

In our proposed method, the watermark is embedded in the chrominance channels (U and V) in a YUV representation of the centre view using the DT CWT, as shown in the block diagram in Fig. 2. Modifying the low-frequency coefficients, which are the coefficients of the higher level of a DT CWT decomposition, of a video frame protects against compression and geometric distortion but is more easily visible to the human eye than changing the high-frequency coefficients.

Since changes in the U and V channels of a video frame are less noticeable to the human visual system than those in the Y channel, the watermark is embedded in a higher DT CWT decomposition level of the U and V channels to ensure it is imperceptible and resistant to lossy compression. Robustness to compression is very important as it distorts the watermark as well as the video signal.

Although the watermark for the U channel, w_u, is a 2D pseudo-random array of elements of 1's and -1's generated using a key, K, for ease of understanding, we show this in Fig. 2 as an image representing the message "copyright". The horizontal and vertical dimensions of the watermark, w_u, are eight times smaller than those of the U channel. The watermark generation key, K, i.e., the watermark pattern, is the same for β consecutive frames. However, this pattern is then never repeated in the remainder of the video sequence. The optimal length of β is a trade-off between robustness to TFA and WER attacks, and is selected experimentally in Section IV-C. The watermark for the V channel, w_v , is generated from w_u and, because of the high correlation between the chrominance channels, it is a 180° rotated version of w_u to avoid false positive errors, as discussed later in Section III-B. Due to the redundancy of the DT CWT, some components of the pseudo-random watermark that lie in the null space of the inverse DT CWT may be lost during the inversion process, hence we embed the level 1 coefficients of a 1-level DT CWT of the watermark rather than directly embedding. It should be noted that, before applying the DT CWT decomposition, both w_u and w_v are up-sampled by a factor of 2.

The watermark is embedded in each sub-band of the level 3 complex coefficients of a 3-level DT CWT of the U and V channels, $H_3^{U}{}_{,d}$ and $H_3^{V}{}_{,d}$ respectively.

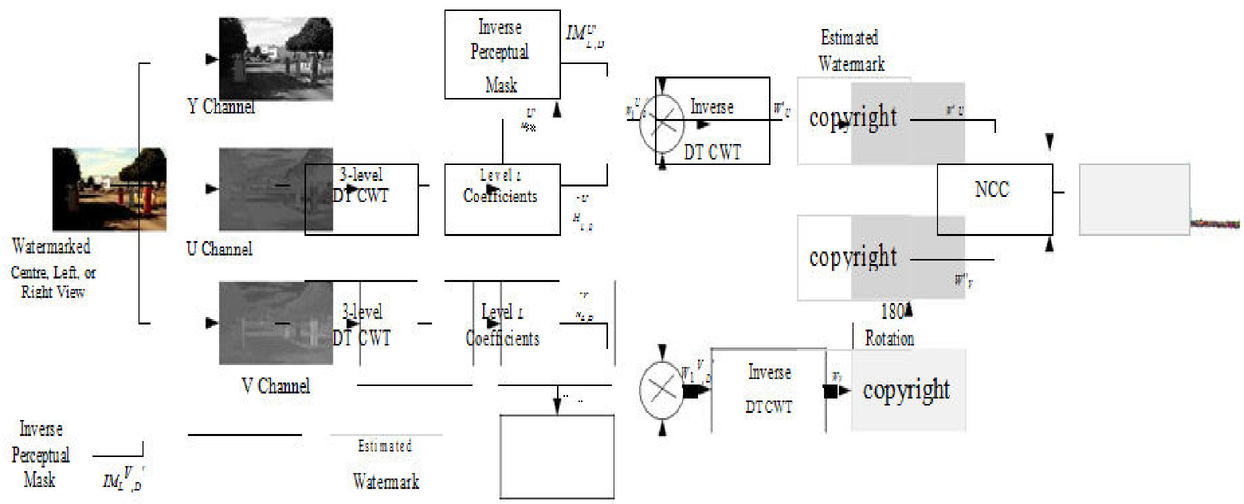


Fig. 3. Watermark extraction from both U and V channels, where any view from the centre, left, and right views can be input to the decoder and l =1, 2, and 3 denote 3 levels of a 3-level DT CWT decomposition.

where d = 1, 2, · · · , 6 are six directional sub-bands of the DT CWT decomposition, the symbol • denotes an element.

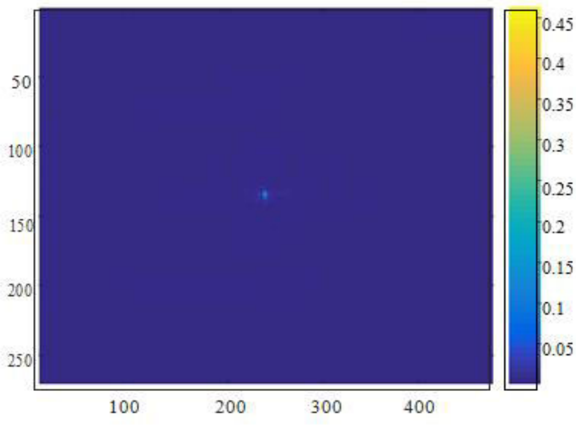
where $W_1{}^{U}{}_{,d}$ and $W_1{}^{V}{}_{,d}$ are the level 1 coefficients of a 1-level DT CWT decompositions of the up scaled versions of w_u and w_v respectively. The factor α, which controls the strength of the watermark, is doubled in the U embedding compared with that in the V embedding because the maximum U sensitivity is about half the maximum V sensitivity. $M_3{}^{U}{}_{,d}$ and $M_3{}^{V}{}_{,d}$ are the perceptual masks for the U and V embedding respectively. A perceptual mask defines the embedding strength of the watermark and ensures the watermark is imperceptible. Each element of the masks controls the embedding strength of the corresponding watermark coefficient. These masks are created from the level 3 coefficients, $H_3{}^{U}{}_{,d}$ and $H_3{}^{V}{}_{,d}$, of a 3-level DT CWT decomposition of the corresponding channel. To create the mask for each of the six sub-bands of level 3 of a channel, a low-pass filter is applied to the magnitude of every sub-band of the level 3 coefficients of the corresponding channel. The resulting arrays are divided by a step value γ which controls the magnitude of the mask coefficients (a larger value of γ produces smaller magnitudes of the mask coefficients.

where L(·) is a 2 × 2 low-pass filter defined as:

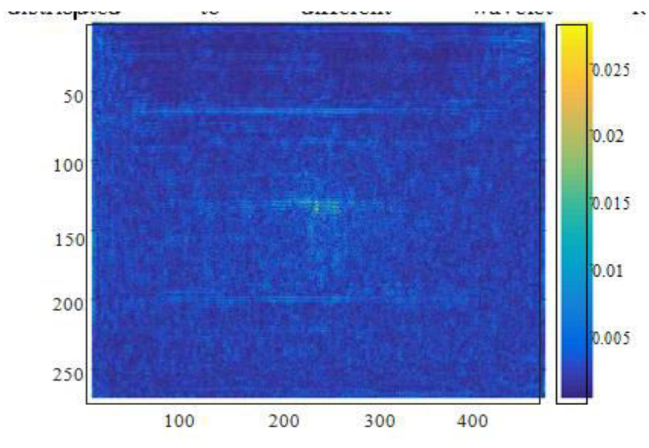
By applying the inverse DT CWT on the watermarked coefficients, a watermarked centre view is obtained which, together with the depth map, is passed through the DIBR system to generate watermarked virtual left and right views. This process is repeated for every frame of a video sequence and results in watermarked centre, left, and right video sequences.

B. Watermark Extraction

Although, during the embedding process, the watermark was embedded in the centre view, this view or either of the virtual views (left or right) might be input to the watermark decoder. The proposed watermark extraction process is shown in Fig. 3. The watermark at the decoder is extracted from any level (1, 2 or 3) of a 3-level DT CWT depending on the downscaled video resolution, although it was embedded in only the level 3 coefficients at the encoder. This procedure is incorporated to mitigate the truncation of the high-frequency bands from a video frame when downscaling occurs. If downscaling to an arbitrary resolution is performed on a frame, the high-frequency coefficients are removed and the watermark coefficients distributed to different wavelet level



(a)



(b)

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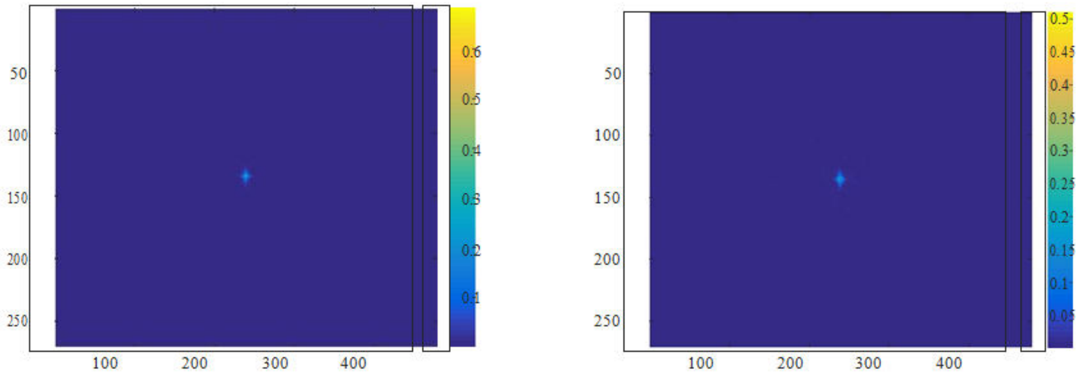

(b) (c)

Fig. 5. Correlation strength when the NCC is performed between the original watermark and the watermark extracted from the (a) U channel and (b) V channel.

Although the real-time execution of the watermark detection filter at the ISP seems computationally complex, a hardware-based implementation of the watermark detection system using a field-programmable gate array (FPGA) could provide a feasible solution.

In the current version of the watermarking scheme, effectively, only a single bit is transmitted using P = 300 video frames. This single bit of information is designed to alert the detection algorithm that a watermark is present. This approach can be extended to embed many more bits of information if more video frames are utilized. This extra information could include authentication information which would then allow video with a non-authenticated watermark to pass through the network. It should be noted that the watermarks are extracted from the both chrominance channels and the NCC is performed between them rather than with the original watermark. Although this technique is useful for access control, it can be extended to the other application like copyright protection if the retrieved watermarks are compared with the original watermark, as shown in Fig. 5. This figure indicates that the watermark decoder is still able to generate the high correlation peaks when the extracted watermarks are compared with the original one. However, in order to do this, same random watermark generation key, which was used during watermark embedding process, is required at the decoder to generate the original watermark patterns. Frame rate conversion breaks the synchronization of the key sequence which results in the watermark detection using the wrong key. For this reason we compare the watermarks retrieved from both channels so that the watermark extraction is not affected by temporal desynchronization, such as frame dropping, frame insertion or frame rate conversion.

CONCLUSIONS

In this paper, we proposed a video watermarking scheme for DIBR 3D video in which the watermark was embedded in the level 3 coefficients of a 3-level DT CWT decomposition of the centre view. We chose both the chrominance channels for embedding the watermark which allowed a much stronger watermark than is possible with the alternative luminance embedding methods while still maintaining an imperceptible difference in quality to the original video. The perceptual quality of the watermarked centre view was verified by a 2D subjective test, and a 3D subjective assessment was also conducted to judge the watermarked 3D video quality and visual comfort. Any view, from the centre or virtual left or right generated from the watermarked centre view and depth map using the DIBR technique, can be the input to the watermark decoder. At this stage, the watermark was extracted from any level of a 3-level DT CWT decomposition rather than from only the embedding level, dependent on the downscaling in video resolution, in order to provide robustness against an arbitrary downscaling in resolution. Due to the shift invariance property of the DT CWT, the watermark was not only robust to the DIBR process but also against geometric attacks such as scaling, rotation, and cropping. The experimental results showed that the proposed method was also effective against H.264/AVC compression, additive noise, baseline distance adjustment, and both 2D and 3D camcording.

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A STUDY ON TWO-STAGE POOLING OF DEEP CONVOLUTIONAL FEATURES FOR CBIR

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ABSTRACT

Convolutional Neural Network (CNN) based image representations have achieved high performance in image retrieval tasks. However, traditional CNN based global representations either provide high- dimensional features, which incurs large memory consumption and computing cost, or inadequately capture discriminative information in images, which degenerates the functionality of CNN features. To address those issues, a two-stage partial mean pooling (PMP) approach to construct compact and discriminative global feature representations. The PMP is meant to tackle the limits of traditional max pooling and mean (or average) pooling. By injecting the PMP pooling strategy into the CNN based patch-level mid- level feature extraction and representation.

Index Terms— image representation, image retrieval, convolutional neural network, feature pooling, compact descriptor

I. INTRODUCTION

Image retrieval has attracted extensive attentions in both academia and industry over the last decade. Feature representation is a core factor influencing both accuracy and efficiency in image retrieval tasks. Traditionally, hand-crafted local features such as SIFT and SURF are aggregated to a global representation by methods such as Bag-of-Words (BoW), Vector of Locally Aggregated Descriptors (VLAD) and Fisher Vector (FV). In recent years, with the rapid development of deep learning, the features extracted from pre- trained Convolutional Neural Network (CNN) models have achieved higher performance and flexibility than traditional hand- crafted aggregated descriptors in typical image retrieval tasks (e.g., scene retrieval, landmark recognition, etc).

Generally speaking, the CNN based image representations can be divided into two types. The first type presents whole images to a pre-trained CNN model and get global representations. A simple approach is to extract high level features from fully connected layers such as fc6/fc7 in AlexNet or Caffe Net. However the raw high dimensional CNN features are much less efficient due to time consuming similarity distance computing. Recent work has applied Principal Component Analysis (PCA) to reduce feature dimension further. Although PCA can transform the features to a low-dimensional representation, the transformation matrix is very large , thereby incurring a time-consuming reduction process. Additionally, if convolutional or pooling features are treated as common vectors, the property that each convolutional or pooling feature map is composed by position related responses is ignored, while feature pooling can handle such problem. Therefore, feature pooling based middle layer representations are considered. Babenko et al. proposed sum pooling to reduce the dimension of the last convolutional or pooling layer features and achieved performance improvement. However, image retrieval tasks usually incur a complex scene, involving multiple scales, cluttered background, as well as multiple subjects, which renders it unsuitable or less optimal for the global representations of a whole image to capture necessary semantic information in retrieval tasks.

To deal with the weakness of whole image based global representations, the second type of methods extracts CNN features of image patches from original images and aggregates them into global representations. For example, Gong et al. have proposed a MOP-CNN scheme, which aggregates deep features of sliding windows of different scales using VLAD. However, the VLAD method is limited by dimension curse since the length of global features are C times of patch-level features, where C is the vocabulary size. MOP-CNN uses PCA to reduce feature dimension which still faces the disadvantages of PCA dimension reduction as mentioned above. Some other works prefer simple approaches such as max pooling or mean pooling (also known as “average pooling”) to get compact global image representations. Unfortunately, max pooling is easily affected by extreme responses while mean pooling may incur background distractors, thereby degenerating meaningful responses.

To address the drawbacks of previous approaches, we propose an effective and efficient two-stage partial mean pooling (PMP) strategy and embed PMP into an advanced feature extraction framework. The proposed PMP attempts to alleviate the disadvantages of both max pooling and mean pooling. In feature extraction, PMP is applied in two stages: 1. intra-patch pooling to capture the discriminative responses from convolutional feature maps; 2. inter-patch pooling to aggregate patch-level features into compact global representation. Extensive evaluation shows that the proposed PMP is superior to max pooling and mean pooling. Meanwhile, the proposed feature extraction framework significantly improves the state-of-the-art retrieval accuracy on several benchmark datasets, with a fairly short feature dimension.

The rest of the paper is organized as follows. In Section 2, we introduce the intra-patch and interpatch pooling strategy, as well as the feature extraction framework. Extensive experiment results and comparison analysis are presented in Section 3. Finally, we conclude this paper in Section 4.

II. FEATURES EXTRACTION APPROACH

In this section, the proposed PMP based feature extraction frame- work (See Fig.1) is described in detail. Our framework includes three main stages: patch detection, mid-level feature extraction and two-stage partial mean pooling.

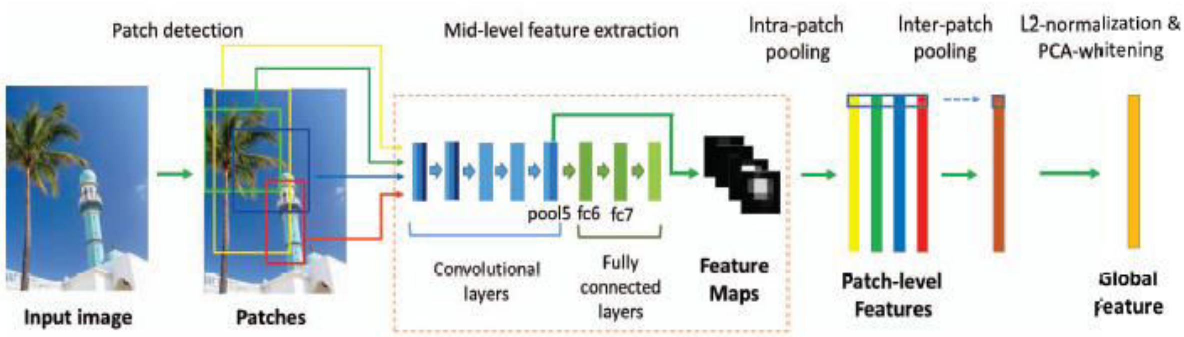

Fig. 1. Overview of the proposed feature extraction framework. The pool5, fc6 and fc7 layers are indicated for CaffeNet. For VGGNet, the network is deeper but remains similar structure.

Patch Detection

As aforementioned, presenting whole images to a pre-trained net- work can hardly deal with the problems of multiple scales, noisy background and abundant subjects. Inspired by recent successful object detection approach using R-CNN, we apply object proposal algorithms to detect regions with high objectness followed by region based feature extraction. In this way, the interference effects of background and other distracting objects can be reduced at the patch level. As detected patches are those regions with high objectness and CNN features are good at describing the semantic information of objects, the CNN mid-level features will be extracted for each patch and then aggregated to form a global representation.

As analyzed in, among most recent object proposal algorithms, BING is with the lowest computation complexity (the whole detection process only takes 10ms in a single thread over a normal PC with 2.6GHz CPU). Towards high efficient feature extraction, we adopt BING as the patch detection algorithm.

It is worthy to mention that MOP-CNN has proposed to handle scale variance by sliding windows with different scales. However, since the patches are greedily detected at pre-determined scales and sliding steps, there is no guarantee that those patches can capture meaningful objects in appropriate sizes. For example, the parts of two distinct objects may be covered in a single image patch, which yields inferior CNN features. Thus our work does not prefer MOP-CNN.

Mid-level Feature Extraction

As illustrated in Fig.1, given a pre-trained CNN model, we extract the last convolutional/pooling feature maps for each image patch as mid-level feature. Specifically, the pool5 layer features from are considered in our work. We propose to sort the responses of pool5 layer from Caffe Net or VGGNet in a descending order within each feature map. As detailed below, our experiment study has shown that the sorted features provide better description than raw features.

Intuitively, the sorted mid-level features avoid the explicit hard coding of location information and thus handle position variance better than raw features. Sorting can gather higher responses for meaningful objects at different locations, so that the similarity distance computing is much less affected by object location variance. For example, each of 256 feature maps in the pool5 layer in is in the form of a 6×6 matrix, of which each matrix element actually codes information of a distinct position. To capture the structure information of pool5 layer, we could successively concatenate the responses of 36 elements, while the position invariance would not be satisfied. Thus, our mid-level feature extraction sorts the responses to make the feature less sensitive to position variance.

In addition, compared to the memory and time consuming fully connected feature, the use of convolutional or pooling layer can significantly reduce the memory and computation cost because fewer large matrix multiplications are involved in feature extraction.

However, the dimension of sorted mid-level features are still too high for retrieval. Hence, we introduce the two-stage partial mean pooling to construct discriminative and compact features in the following.

Two-stage Partial Mean Pooling

The two stage partial mean pooling (PMP) is then applied to the patch-level features for generating compact representations.

Intra-patch Pooling. Intra-patch pooling is to capture the discriminative responses on the feature maps and transform the feature to a low-dimensional representation. As a sort of quantization, the proposed PMP based intra-patch pooling can remove the negative effects of position variance in feature representation. We formulate the PMP for intra-patch pooling as follows.

Fig. 2. Response visualization by the feature maps of pool5 layer using Caffe Net. Each feature map is supposed to represent a concept. A moderate number of response values may illustrate a concept meaningful feature map.

Note that PMP may degenerate to the form of traditional max pooling or mean pooling when K1= 1 or h × w. PMP is meant to seek for a trade-off between max pooling and mean pooling by considering the largest K1 responses, in which, by appropriately setting K1 value, better pooling results of feature maps can be expected. As shown in Fig.2, when K1 is set to an appropriate value, say, of h×w, the pooling result of PMP shows much better delineation of a meaningful pattern (“donkey”, “leaf”, etc.). By contrary, max pooling tends to capture noisy strong response values when the majority of responses are mild, while mean pooling would yield a skewed value when the meaningful responses are just from the minority.

Inter-patch Pooling. In inter-patch pooling stage, the features of different patches are aggregated to form a global representation. Rather than max or mean pooling, PMP is applied to aggregate the features of different patches as well. Fig.3 illustrates different pooling effects of max, mean and PMP methods. Note that we do not apply the typical aggregation approaches such as VLAD and FV as they incur high-dimensional representations. Actually, recent aggre- gation N denote work the has number preferred of patches simple but and effective Y ¯pooling [15, 16]. Let denote the features of all patches after intra-patch pooling:

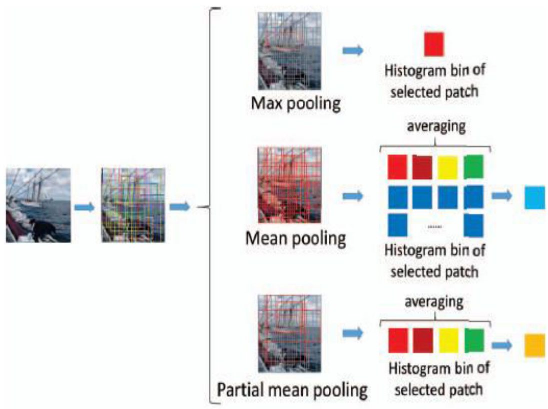

Fig. 3. A toy example of inter-patch pooling to compare max, mean and partial mean pooling (note the top 50 patches are shown). Each block indicates a response value. The block colors (from blue to red) indicate different response strength (from weak to strong).

The PMP provides more discriminative pooling results for "ship" object. Compared with mean pooling, PMP discards noisy back- ground patches. Compared with max pooling, PMP can better handle scale and position variance by averaging the features of meaningful patches.

III. CONCLUSION

We have studied a two-stage partial mean pooling strategy towards an advanced CNN feature extraction framework. The compact and discriminative image representation outperforms state-of- the-art methods. How to incorporate low-level invariant features into this feature extraction framework (i.e., the effective combination of CNN and SIFT features) will be included in our future work.

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FULL CODE RATE COMPLEX NON-ORTHOGONAL STBCS FOR THREE TO SIXTRANSMIT ANTENNAS

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Abstract— **Alamouti is Pioneer of Complex Orthogonal STBCs for two transmit branch diversity with code rate one. Later Tarkoh investigated for four to nine transmit antennas with code rate 1/2. Jafarkhni proposed for non orthogonal STBCS four transmit antennas with code rate one. The theory of space-time block codes was further developed by Weifen Su and Xian-Gen Xia for 5 and 6 transmit antennas code rate of 7/11 and 3/5. O. Tirkkonen and A. Hottinen Proposed space-time block codes for eight transmit antennas of code rate 1/2. Recently five, six, seven and eight transmit antennas generalized complex orthogonal space-time block codes of code rate with 2/3, 2/3,5/8, and 5/8 proposed by Liang. This paper intends new matrix non-OSTBC for three to six transmit antennas with full code rate.**

Keywords— non-OSTBC, fading, code rate

INTRODUCTION

The mid 1970's orthogonal designs were first introduced by A. V. Geramita, J. M. Geramita, and J. S.Wallis , but mathematical results in other background dating from the 1890s and the 1920s laid the establishment for these combinatorial structures. The major significance of orthogonal designs which acts as transmission matrices in space-time wireless communications can be real or complex and square or rectangular matrices.The orthogonal designs have two classes of STBCs. The real orthogonal designs for real constellations such as PAM are considered as one class. These codes have been well developed by A. V. Geramita, J. M. Geramita and J. S.Wallis. The complex orthogonal designs for complex constellations such as QAM and PSK are grouped as the other class. These codes or complex orthogonal designs or Hermitian compositions of quadratic forms are not well understood as one understands the STBCs from real orthogonal designs.

2. LITERATURE REVIEW

Currently, MIMO technology has attracted great consideration because of its great advantages. MIMO considered that the capacity of wireless channel can be Significantly increased by the use of multiple antennas at transmitter and receiver and mitigating the multi channel fading[1]. One of the main element in MIMO system, called space-time block coding(STBC), can improve the reliability of MIMO system. A noteworthy coding scheme named Alamouti was invented in 1998 for MIMO system, which has full transmitted diversity and can be decoded by maximum-likelihood method due to its orthogonality [2,3]. Yet, the code rate of an orthogonal STBC (OSTBC) is upper bounded by 3/4 transmitted symbols per channel use for more than two transmit antennas [4]. As a result by relaxing orthogonality constraint, quasi-orthogonal STBC (QOSTBC) with code rate one and transmitted symbols per channel use is attained at the cost of higher decoding complexity [5].The space-time block codes exist with symbol transmission rates 7/11 and 3/5, respectively, from General Complex orthogonal designs with linear processing [7] for 5 and 6 transmit antennas. To show high rate OSTBCs of system design& Computer aided method for any number of transmit antennas is presented in [8].Similarly high rate OSTBCSfor 2k-1 or2k transmit antennas with code rates of$(k+1)/(2k)$ is found in [9]. X.B.Liang represented new OSTBC for eight transmit antennas by padding eight column to existing 7 transmit antennas. O. Tirkkonen and A. Hottinen [6] Proposed space-time block codes for eight transmit antennas of code rate ½.The present OSTBC for eight transmit antennas achieve same maximal rate 5/8 as well as same delay 56 of 7 transmit antennas [10].The theory of space-time block codes was further developed by M.A.Islam Jewel, M.Rahman[11]. They defined space time block codes in terms of orthogonal code matrices. The properties of these matrices ensure full rate for four transmit antennas. We proposed full rate for eight and sixteen transmit antennas [12][13] . O. Tirkkonen and A. Hottinen Proposed space-time block codes for eight transmit antennas with code rate ½. X.B.Liang [10] has reported OSTBC for ten transmit antennas, which transmit 252 information symbols in 420 time slots, a code rate of 3/5 resulted.

3. METHODOLOGY

The non-OSTBC three to seven transmit antennas matrices obtained full code rate in which number of symbols, number of time intervals is n(n-1) /2and number of Alamouti's matrices is n(n-1)/4. Here n represents number of transmit antennas.

3.1 Existing Four to Six Transmit Antennas

S.No	Number of Transmit Antennas	Symbols (k)	Delays (p)	Code rate (k/p)
1	3	3	4	3/4
2	4	3	4	3/4
3	5	10	15	2/3
4	6	20	30	2/3

3.1 Proposed Four to Six Transmit Antennas

Constructing G_3 as a 4x3 OSTBC matrix is expressed as eq 1.1 where code rate R = 4/4 = 1, and the block length p =4

$$G_3 = \begin{bmatrix} s_1 & s_2 & 0 \\ -s_2^* & s_1^* & 0 \\ 0 & s_3 & s_4 \\ 0 & -s_4^* & s_3^* \end{bmatrix} \qquad 1.1$$

Another 6x4 non-OSTBC G_4 is constructed as given below as eq 1.2

$$G_4 = \begin{bmatrix} s_1 & s_2 & 0 & 0 \\ -s_2^* & s_1^* & 0 & 0 \\ 0 & s_3 & s_4 & 0 \\ 0 & -s_4^* & s_3^* & 0 \\ 0 & 0 & s_5 & s_6 \\ 0 & 0 & -s_6^* & s_5^* \end{bmatrix} \qquad 1.2$$

The proposed five transmit antennas which can send 8 information symbols with the block length p=8 have the code rate 8/8=1, and the matrix is expressed as eq 1.3

$$G_5 = \begin{bmatrix} s_1 & s_2 & 0 & 0 & 0 \\ -s_2^* & s_1^* & 0 & 0 & 0 \\ 0 & s_3 & s_4 & 0 & 0 \\ 0 & -s_4^* & s_3^* & 0 & 0 \\ 0 & 0 & s_5 & s_6 & 0 \\ 0 & 0 & -s_6^* & s_5^* & 0 \\ 0 & 0 & 0 & s_7 & s_8 \\ 0 & 0 & 0 & -s_8^* & s_7^* \end{bmatrix} \qquad 1.3$$

The proposed six transmit antennas which can send 10 information symbols with the block length p=10 have the code rate 10/10=1, and the matrix is expressed as eq 1.4

$$= \begin{bmatrix} s_1 & s_2 & 0 & 0 & 0 & 0 \\ -s_2 & s_1^* & 0 & 0 & 0 & 0 \\ 0 & s_3 & s_4 & 0 & 0 & 0 \\ 0 & -s_4 & s_3^* & 0 & 0 & 0 \\ 0 & 0 & s_5 & s_6 & 0 & 0 \\ 0 & 0 & -s_6^* & s_5^* & 0 & 0 \\ 0 & 0 & 0 & s_7 & s_8 & 0 \\ 0 & 0 & 0 & -s_8^* & s_7^* & 0 \\ 0 & 0 & 0 & 0 & s_9 & s_{10} \\ 0 & 0 & 0 & 0 & -s_{10}^* & s_9^* \end{bmatrix} \qquad 1.4$$

CONCLUSIONS

For conventional complex codes of number of transmit antennas (five to six), the number of symbols and time delays is very large. The above problem is alleviated by using non-OSTBC concept reducing the number of symbols and time delays as shown in Table 1.2 which will reduce the complexity of the system. For conventional complex codes of number of transmit antennas (three to four) achieve less full code rate as compared to proposed systems which will transmit more information carrying capacity.

Table 1.2 Proposed Non-OSTBCs for Three to Seven Transmit Antennas

S.No	Number of Transmit Antennas	Symbols (k)	Delays (p)	Code rate (k/p)
1	3	4	4	1
2	4	6	6	1
3	5	8	8	1
4	6	10	10	1

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LOCAL BINARY PATTERN BASED MOTIF SHAPE PATTERNS FOR TEXTURE CLASSIFICATION

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Abstract— **The present paper proposes a method for extracting features based on motif shape patterns on Local Binary Pattern for effective stone texture classification. In this method Local Binary Pattern (LBP) is computed on the image and then motif shape parameters are evaluated. These shape patterns are used as features for classification. The present method experimented on the Dataset consists of various brick, granite, and marble and mosaic stone textures collected from Vistex, Mayang database and also from natural resources from digital camera. The experimental results and comparison with the other methods show the outperforming of the proposed method with the existing methods.**

Keywords—Texture classification, LBP, Motif shape patterns

I. INTRODUCTION

Texture is a surface property which gives combined information on the smoothness, coarseness, and regularity of objects. On digital images, it reflects as local variations of the gray-scale content. The typical automatic texture classification system involves two steps: (1) a feature extraction step, where a set of texture features are extracted from the image under study and (2) a classification step, where a texture class membership is assigned to it according to the extracted texture features.

Texture classification plays an important role in computer vision and image processing. In the past decades, numerous algorithms for texture feature extraction have been proposed, many of which focus on extracting texture features that are robust to noises, rotation and illumination variants [1]. Goyal et al. [2] proposed a method by using texel property histogram. Cohen et al. [3] characterized texture as Gaussian Markov random fields and used the maximum likelihood to estimate rotation angles. Chen and Kundu [4] addressed rotation invariant by using multichannel sub-bands decomposition and hidden Markov model (HMM). Recently, Varma and Zisserman [5,6,7] proposed to cluster a rotation invariant texton dictionary from a training set, and then form the textural histogram based on these textons. Later, Xu et al. [8,9,10] presented scale invariant texture classification methods by using a multi-fractal spectrum (MFS). In [11, 12], Ojala et.al. proposed to use the Local Binary Pattern(LBP) for rotation invariant texture classification. LBP is a simple yet efficient operator to describe local texture, and has been proven to be invariant to monotonic grayscale transformations.

The primitives and their spatial arrangements are used to characterize textures. For example, morphological operations are used to characterize textures [13]. Song's method [14] decomposes textures into a set of scale images, finds square texels of the same size at each scale, and uses the histogram of the texels as texture features. The method proposed by Gui et al. [15] extracts the size, position, periodicity, and spatial organization of texels to analyze textures. Khellah's method [16] uses the similarity between pixels and their surrounding neighbors within a predefined window and generates a global map called the "Dominant Neighborhood Structure". The present paper evaluates motif shape patterns on LBP of textures. The present paper is organized as follows. In section II, proposed methodology is described, section III includes results and discussions and conclusions are drawn in section IV.

II. METHODOLOGY

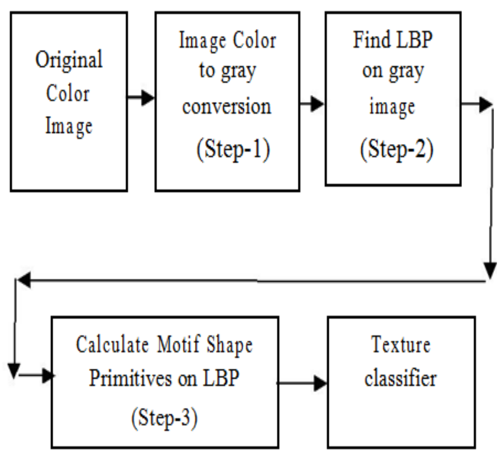


Fig.1: Texture Classification using Motif Shape Patterns on LBP images.

In this proposed method the color image is converted to gray image, on this image the Local Binary Pattern (LBP) is computed and then shape parameters are evaluated for age classification and this method is illustrated in figure 1.

Step -1: Color image to gray image conversion

The given color image is converted into a grey level image using RGB color quantization method.

Step-2: Local Binary Pattern

The LBP is computed on the image for obtaining local neighborhood information of pixels [11]. The computation of LBP with an example is illustrated in figure2. A 3×3 neighborhood consists of a set of nine elements, P = {pc, p0, p1,…, p7}, where p_c represents the gray level value of the central pixel and pi (0≤i≤7) represent the gray level values of neighbor pixels. Each 3×3 neighborhood then can be characterized by a set of binary values bi (0≤i≤7) as given in equation1.

$$b_i = \begin{cases} 0 & \Delta p_i \geq 0 \\ 1 & \Delta p_i < 0 \end{cases} \qquad (1)$$

where $\Delta p_i = p_i - p_c$.

For each 3×3 neighborhood, a unique LBP is derived from the equation2.

$$LBP_{P,R} = \sum_{i=0}^{i=7} b_i \times 2^i \qquad (2)$$

Every pixel in an image generates an LBP code. A single LBP code represents local micro texture information around a pixel by an integer code in between 0 and 255.

Step-3 : Motif Shape patterns

The LBP image is divided into 2 x 2 grids. These grids are then replaced by a particular Peano scan motif which would traverse the grid in the optimal sense. Here, the optimality of the Peano scan is with respect to the incremental difference in intensity along the scan line minimizing the variation in the intensities in a local neighborhood. In general, 24 different Peano scans (motifs) could traverse a 2x2 grid. But we consider only the Peano scans (motifs) which start from the top left a corner of the grid because they represent a complete family of space filling curve, reducing the number of motifs to only six [17]. The motif shape patterns are defined over a 2×2 grid, each depicting a distinct sequence of pixels starting from the top left corner as shown in figure 3 for age group classification and are denoted as Z, N, U, C, Gamma, and Alpha respectively. The present paper considers motif shape patterns on LBP of the texture image. The frequency occurrences of all these shape patterns are evaluated on LBP of the image with a 2 x2 grid from left to right and top to bottom in non-overlapped fashion. The process of finding motif shape patterns on LBP is shown in figure 4.

p_0	p_1	p_2		b_0	b_1	b_2		2^0	2^1	2^2		
p_3	p_c	p_4	→	b_7		b_3	X	2^7		2^3	→	LBP
p_5	p_6	p_7		b_6	b_5	b_4		2^6	2^5	2^4		

92	46	54		1	0	0		1	2	4		
85	56	56	→	1		1	X	128		8	→	185
43	86	78		0	1	1		64	32	16		

Fig. 2: Computation of LBP.

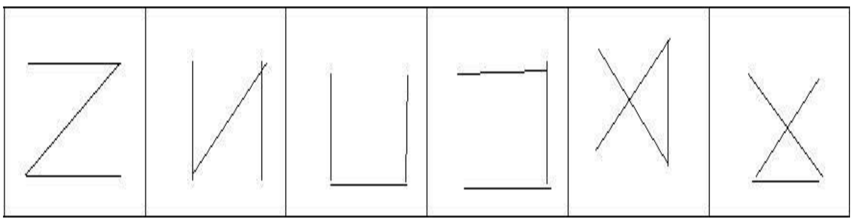
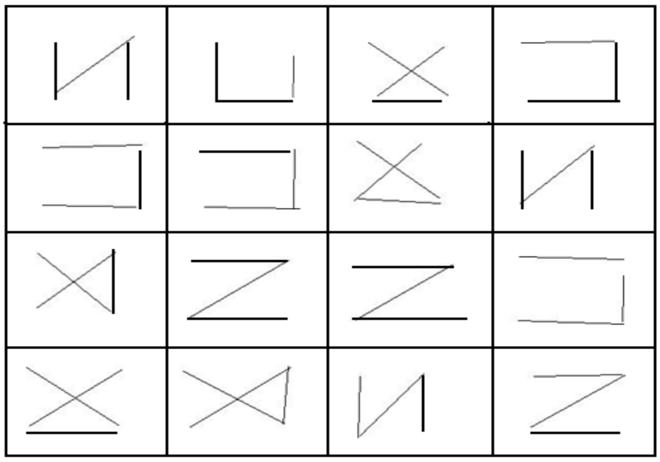


Fig 3: Motif Shape Patterns Z, N, U, C, Gamma and Alpha

205	56	79	65	89	255	255	90
110	48	78	90	187	145	57	64
54	111	219	99	89	74	34	46
210	203	65	90	75	84	38	76
98	73	87	43	67	80	77	87
199	88	200	208	99	105	109	64
97	45	89	79	119	94	154	109
49	99	29	98	97	78	68	255



(a) LBP on texture image

(b) Motif shape patterns on 2x2 grid of LBP

Fig 4: Computation of Motif Shape Patterns on LBP images

III. RESULTS AND DISCUSSIONS

Experiments are carried out to demonstrate the effectiveness of the proposed method for stone texture classification. The present paper carried out the experiments on the dataset, which consists of various brick, granite, marble and mosaic stone textures collected from Vistex, Mayang, Akarmarble, Brodatz databases and also from natural resources from digital camera. Sample of Granite, Mosaic, Marble, Brick textures are shown in Fig.7. In this method, texture images are classified based on frequency occurrences of the shape patterns Z, N, C, U, GAMMA, and ALPHA and Sum of Frequency Occurrences of the Motif

Shape Patterns (SFOMSP). The frequency occurrences of Z, N, C, U, GAMMA, and ALPHA are represented with FOZ, FON, FOC, FOU, FOGAMMA, and FOALPHA respectively. The proposed method is investigated on considered dataset and the results are shown in Table 1. From the results, frequency occurrences motif shape patterns Z, N, U, ALPHA and also sum of frequency occurrences of the motif shape patterns are considered as features for texture classification. The algorithm1 is proposed with these features. This algorithm has given 97% correct classification rate for classifying textures.

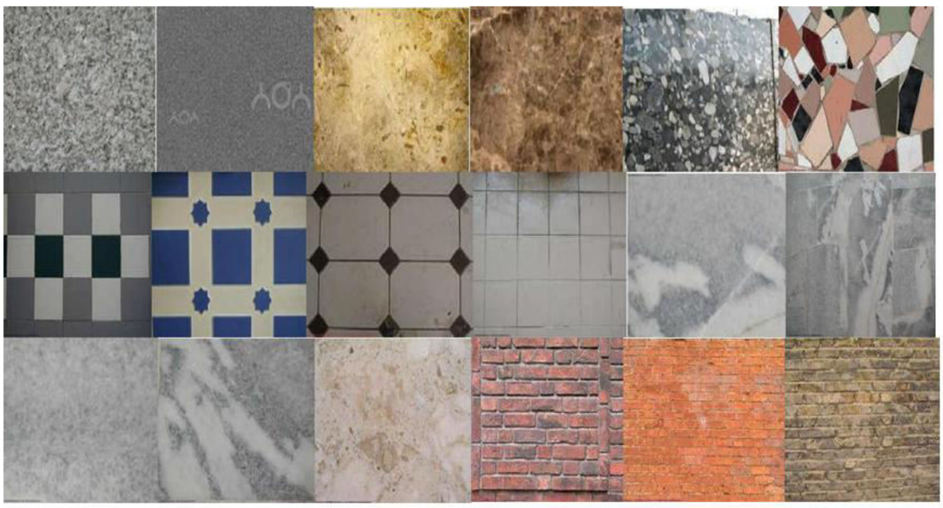


Fig 5: Sample images of Granite, Mosaic, Marble, Brick Textures

Table 1: The Frequency Occurrences of Motif Shape Patterns on LBP Images

IMAGE	FOZ	FON	FOC	FOU	FOGAMMA	FOALPHA	SFOMSP
Granite1	1274	1379	1597	1759	971	841	7821
Granite2	911	1829	1242	1924	996	881	7783
Granite3	1143	1651	1505	1873	1017	845	8034
Granite4	1228	1607	1627	1944	958	832	8196
Granite5	1118	1594	1589	2005	949	986	8241
Granite6	1421	1362	1751	1659	949	832	7974
Mosaic1	1956	2404	2353	2682	1604	1507	12506
Mosaic2	3558	4096	3665	4135	2211	2328	19993
Mosaic3	3606	3171	3803	3308	2082	2111	18081
Mosaic4	3141	3013	3431	3311	1931	1872	16699
Mosaic5	3628	3361	3834	3536	2140	2128	18627
Mosaic6	2331	2141	2668	2507	1158	1208	12013
Marble1	2005	1870	2398	2264	1133	1028	10698
Marble2	1979	1932	2325	2382	1039	960	10617
Marble3	2032	1487	2486	1994	851	916	9766
Marble4	2200	1848	2577	2279	1186	1157	11247
Marble5	1837	1676	2219	2084	972	963	9751
Marble6	1622	1560	2118	2009	908	855	9072
Brick1	2205	1220	2731	1570	777	776	9279
Brick2	2229	1038	2616	1432	709	804	8828
Brick3	2426	1042	2793	1541	736	701	9239
Brick4	2315	1137	2763	1635	839	841	9530
Brick5	2493	1281	2910	1806	712	744	9946
Brick6	2420	1262	2688	1703	713	722	9508

Algorithm 1: Texture classification using frequency occurrences of Motif Shape Patterns on LBP images

BEGIN

IF (FOZ>910)&(FOZ <1425)

&(SFOMSP> 7745) &(SFOMSP< 8245))

PRINT (“GRANITE”) ;

ELSE

IF (FOALPHA > 1470)&(FOALPHA <2330)

&(SFOMSP> 12010) &(SFOMSP< 20000))

PRINT (“MOSAIC”);

ELSE

IF (FOU> 1990)&(FOU<2290)

&(SFOMSP> 8970) &(SFOMSP< 11250))

PRINT (“MARBLE”);

ELSE

IF (FON> 800)&(FON<1300)

&(SFOMSP> 9200) &(SFOMSP< 10000))

PRINT (“BRICK”);

END

The proposed method for texture classification is compared with existing methods syntactic pattern on 3D method [18], primitive pattern unit approach [19] and texton feature evolution method [20]. The percentage of classification rates of the proposed method and other existing methods are listed in Table 2. From the Table 2, it clearly indicates that the proposed LBP based motif shape patterns method outperforms the other existing methods. The comparison chart of the proposed method with the other existing methods is shown in Fig.6.

Table 2: Mean % classification rate of the proposed and existing methods.

Image Dataset	Syntactic Pattern on 3D method [18]	Primitive Pattern Unit approach [19]	Texton Feature Detection [20]	Proposed method
Mayang	93.29	92.19	95.56	96.15
VisTex	92.53	92.56	94.15	96.05
Akarmarble	93.30	91.29	95.27	97.12
Brodatz	93.59	92.16	94.97	95.85

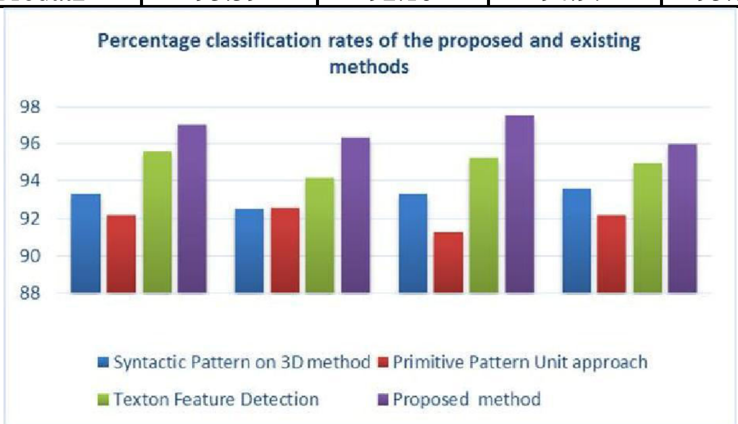


Fig 6: Classification rate of the proposed and existing methods

IV. CONCLUSION

The present paper developed a method for texture classification using motif shape patterns on LBP of images. This proposed method extracts the local information of each pixel of an image using LBP and then the motif shape patterns are evaluated as features for classifying textures. The algorithm1 with these new set of features classified texture images with a good classification rate compared to other existing methods. This method is very simple, efficient and accurate for texture classification. The LBP variants with different shape parameters and textural properties as features can be extended as future work.

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A STUDY ON INDUSTRIAL APPLICATIONS OF OPTICAL CHARACTER RECOGNITION - DIGITIZING RECORDS

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ABSTRACT: Organizations across all industries are in the process of digitizing records such as patient records, client forms, billing records, maintenance records, electronic health records (EHR) to reduce storage costs, increase information security and enable downstream analytics processes. Nevertheless, when organizations transition from paper to electronic copies, their legacy data in paper format needs to be scanned and made intelligible to analytics algorithms. Optical Character Recognition (OCR) is the process by which scanned copies of records are analyzed, characters recognized, and made available for editing, search and analytics. OCR involves a series of advanced Computer Vision and Pattern Recognition techniques. In this paper we have presented the structure of the OCR system in the first part , Working in the second part and its applications in the last part.

Keywords:De-Skewing,De-noising, Character Enhancing, Histogram Equalization

INTRODUCTION

Organizations across all industries are in the process of digitizing records such as patient records for healthcare, billing records, maintenance logs or clients forms. After digitization, the cost of storage and communication is greatly reduced; information security is also greatly improved, especially in the context of confidential information. In a digital format, a document may be shared instantaneously to all the relevant parties, without the need to manage paper file rooms. Furthermore, when digital documents are available in a text-searchable format, their content can be easily searched for specific information without the need to spend time combing through a large volume of paper documents. In addition, the recent years have seen a dramatic drop in the cost of storage, thus giving the upfront costs of digitization (scanning, OCR and filing) a positive Return-On-Investment.

Furthermore, digital records are the foundation of further downstream analytics. These analytics programs are especially important for large organizations such as Healthcare Provider and Payer [1] to respond to the industry trends such as data integration, consumerization, patient risk stratification, value-based care and cost reduction. Analytics services to healthcare providers address challenges such as reducing the re-admission rates to maximize revenues and patient satisfaction, moving to value-based reimbursements through the monitored implementation of specific quality measures and deriving actionable insights with data integrated from multiple sources. Analytics services to healthcare payers address challenges such as stratifying the members' risk and identifying opportunities for gap closure through preventive measures, fraud prevention, and cost optimization.

When organizations transition from paper to digital record systems, one of the first steps is to scan the available legacy data in paper format. However, simply doing so does not render the text intelligible for algorithms to edit, search or analyze. Rather, these scans are simple photographic images of the text, rather than the text itself. To make the text intelligible to algorithms is a prerequisite before further analytics.

Modern forms are designed in such a way to make it easy for OCR systems to analyze scanned images of these forms and turn them into machine-encoded documents. These forms typically include "comb-fields" to limit the position and spacing of the characters, and check boxes when a limited number of choices are available. Other techniques such as 1D or 2D code bars are becoming increasingly popular, along Quick Response Codes (QR Code, ISO/IEC 18004) [Figure 1] OCR consists in analyzing the scanned images, detecting the characters on that image, and transforming the text image into machine-encoded characters, words and sentences.

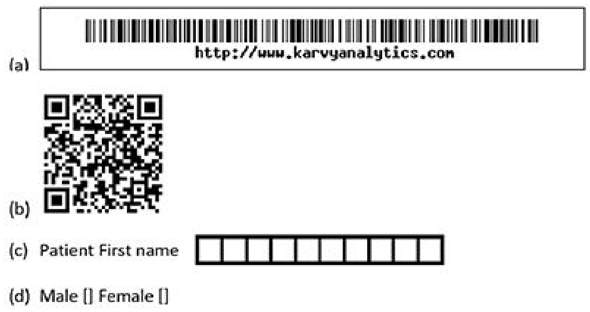

Figure 1 Example of tools used to optimize the digitization of forms. (a) 1d code bar, (b) QR code, (c) comb-field text field, (d) check boxes

Structure of OCR systems

OCR systems have been investigated since 1920s and leverage research fields such as computer vision, pattern recognition and artificial intelligence. The typical OCR process can be splitinto three fundamental steps (image pre-processing, character recognition, and post-processing correction), which are detailed below:

Pre-processing a scanned image for OCR

Although typical commercial OCR systems report a character recognition success rate of 90- 99%, much of the performance depends on the scanning process and preparation of the image before the actual character recognition is performed. It is therefore, imperative to optimize the scanning process itself in order to maximize OCR efficiency. The scanning device should be cleaned and maintained regularly, and all scanned documents should have the same orientation.

After a paper document is digitally scanned, the resulting image is processed through a series of algorithms based on Computer-Vision techniques. These pre-processing steps aim to correct irregularities from the scanning process as much as possible.

- **De-Skewing:** a process by which the bounding box of the scanned document is detected, and the image is rotated such that the document may present itself following a typical upright position, as most forms follow a standard A4 format.[Figure2]

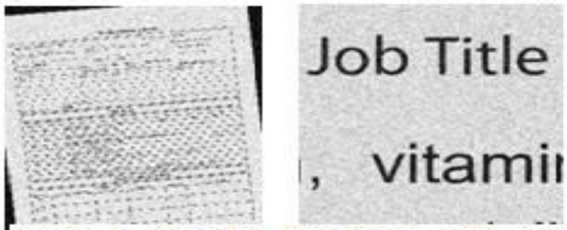

Figure 2 Examples of skewed (left) and relatively noisy (right) scanned images

- **De-noising:** a process by which 'noise' coming from scanning devices is reduced or eliminated. Noise may present itself under a variety of forms such as pepper-and-salt noise, dark lines and other image artifacts from the scanning device.
- **Character enhancing:** de-noising algorithms may lead to some characters losing their edges. It may therefore become necessary to enhance the characters' edges by applying a sharpening image mask such as a gradient edge filter.
- **Histogram Equalization:** depending on the scanning device, some part of the image may be more exposed than others. This explains why some parts of the resulting document appear brighter than others . Histogram equalization is a technique that aims to balance out the brightest and darkest regions of an image.

Page Segmentation: if the scanned image contains a region that is outside the document, that region should be detected and segmented, as it could lead to additional errors during character recognition

- **Page Layout Analysis:** With the generalization of Big Data infrastructures, record digitization projects now often involve millions of documents formatted in hundreds, sometimes thousands, of different layouts. These layouts may include not only text but also pictures, schemas and charts. To further improve the accuracy of the line-word-character segmentation, the documents are automatically classified based on their layouts[4].
- **Line-Word-Character segmentation:** OCR systems work best when individual characters are identified and isolated. To do so, pattern recognition is applied to detect text blocks, followed by text lines, and finally individual characters shown in figure 3 below.

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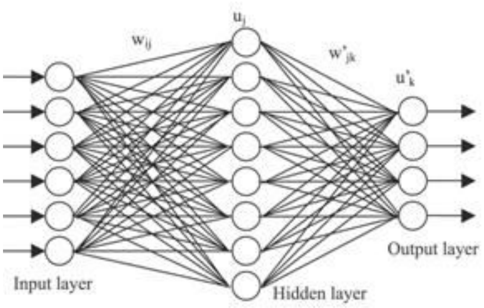
Text blocks are segmented, before individual words and characters.

Character Recognition

Once the position of an individual character in the scanned document is known, pattern recognition techniques are applied at the characters ‘position to correctly identify that the character in a known alphabet. These pattern recognition techniques may be divided into feature-based and feature-less techniques.

The feature-based techniques rely on explicit characteristics of the character such as vertical/horizontal lines, loops, and line intersections. The features of a specific character are compared with a set of features from a known set of characters to identify the most similar character. Feature-based techniques have been used for handwritten documents where the characters’ appearance may vary greatly depending on the document’s author. The list of typical algorithms includes k-nearest neighbors, SVM and Neural Networks [2].



The featureless techniques have gained popularity with documents which use a consistent set of character fonts, or when the scanning process and pre-processing steps successfully eliminate inconsistencies in the image. These techniques consist in identifying the most likely character solely based on the grey-scale value of the character block.

Post-Processing Corrections

Even the best OCR algorithms still produce errors. For instance, a 99% performance, or 1 error in 100 characters, may lead to unnecessary costs or information lost. Typical errors include substitution errors (e.g. incorrectly identifying an ‘m’ for an ‘n’), deletion errors (e.g. skipping a character within a word), and insertion errors (e.g. adding a character that was not part of the original text). For that reason, post-processing corrections are often times performed after character recognition. Short of hiring a team of transcriptionists to manually check and correct the documents, the post-processing corrections have to be automated. The corrections typically consist of a spell-check based on a specialized dictionary, such as a language-specific medical dictionary, or other context-based lexicon adapted to the specific text being digitized.

A dictionary-based approach replaces the words by the most likely words in a dictionary. The measure of likelihood is defined using a distance metric between words, such as the Damerau- Levenshtein distance [6]. However, purely dictionary-based corrections do not account for context information contained in the sentence. For instance, “Medical Mÿstory” may be corrected into “Medical Mystery”, as “Mystery” is the most likely word for “Mÿstory” when analyzed out of context. Nevertheless, a context-based correction should be able to point to “Medical History” as the correct answer.

Context-based approaches typically rely on Statistical Language Modeling (SLM), a Natural Language Processing (NLP) technique that aims at modeling regular expressions and word co-occurrences. In the previous example, a Provider- or Payer-specific lexicon should clearly point to “Medical History” as the correct answer. A co-occurrence matrix is defined as the affinity between a word and a regular expression. In our example, “History” would be more strongly associated with “Medical History” than “Mystery”. A co-occurrence matrix thus, models a statistical distribution of word associations in regular expressions. That model is then used by optimization techniques, such as Pointwise Mutual Information[7], to correctly assign the most probable word given the context of a given sentence.

Big Data testing and validation

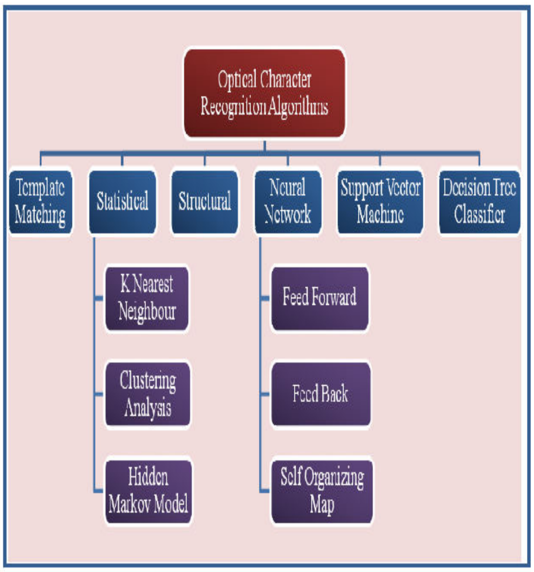
Even the most accurate OCR systems do not perform at 100% accuracy, and a precise estimate of the accuracy level is necessary before engaging further resources in downstream analytics. However, such an estimate may be quite costly to obtain, in part because of the huge size of the datasets, the variety of the digitized documents and the fact that sometimes the output data may be unstructured.

For that reason, a testing and validation automation frameworkhas developed to perform a statistical evaluation of the OCR accuracy level stratified based on the document layout classification. This ensures that the documents with the least accurate OCR output may be further processed to avoid unnecessary costs and errors in downstream analytics.

Optical Character Recognition Algorithms

The following are some of the optical character recognition algorithms

- Template Matching Algorithm
- Statistical Algorithm
- Structural Algorithm
- Neural Network Algorithm
- Support Vector Machine
- Decision Tree Classifier



Among all of them we are presenting only statistical algorithms in this paper.

The purpose of the statistical algorithms is to determine to which category the given pattern belongs. By making observations and measurement processes, a set of numbers is prepared, which is used to prepare a measurement vector. Statistical algorithm uses the statistical decision

functions and a set of optimality criteria which maximizes the probability of

the observed pattern given the model of a certain class.

Statistical algorithms are mostly based on three major assumptions:

I. Distribution of the feature set.

II. There are sufficient statistics available for each class.

III. Collection of images to extract a set of features which represents each distinct class of patterns.

There are two approaches of statistical algorithm.

I. Non-parametric Recognition

In Non-parametric Recognition, a priori data or information is not available.

II. Parametric Recognition

Since a priori data or information is available about the characters in the

training data, it is possible to obtain a parametric model for each character.

The major statistical methods applied in the character recognition field are K

Nearest Neighbor, clustering Analysis, Hidden Markov Modeling etc.

K-Nearest Neighbour Algorithm

The k-Nearest Neighbors algorithm (k-NN) is a non-parametric method usedfor classification. The input consists of the k closest training examples inthe feature space. In k-NN classification, the output is a class membership. Anobject is classified by a majority vote of its neighbors, with the object beingassigned to the class most common among its k nearest neighbors (k is apositive integer, typically small). If k = 1, then the object is simply assigned toclass of that single nearest neighbor [12]. The idea behind k-Nearest Neighboralgorithm is quite straightforward. To classify a new character, the systemfinds the k nearest neighbors among the training datasets, and uses the

Categories of the k nearest neighbors to weight the category candidates.

The k-NN algorithm can be described using the following equation:

$$y(d_{i)} = \arg\max_k \sum_{Xj \in kNN} \mathrm{Sim}\,(d_{i,}x_j)\; y(\,x_{j,}c_k\,)$$

Where, *di* is a test character, *xj*is one of the neighbors in the training set,*y(xj,ck)* ∈*{0, 1}* indicates whether xj belongs to class ck, *Sim(di,xj)* is thesimilarity function for *di*. Above equation means the class with maximal sumof similarity will be the winner .The performance of this algorithm greatly depends on two factors, that is, asuitable similarity function and an appropriate value for the parameter k. Thesimilarity function is the Euclidean distance. It is given by below equation.

$$f(x, p^2) = \sum_{i=1}^{N} (x_i - p_i^2)$$

One of the basic requirements for this method to obtain good performance isthe access to a very large database or labeled prototype but searching throughthe whole database to find the nearest objects to a test image is timeconsuming, and has to be done for every character in a document.

Clustering Analysis

The goal of a clustering analysis is to divide a given set of data or objects intoa cluster, which represents subsets or a group. The partition should have twoproperties. Homogeneity inside clusters: the data, which belongs to onecluster, should be as similar as possible. Heterogeneity between the clusters:the data, which belongs to different clusters, should be as different as possible. Thus, the characters with similar features are in one cluster. Thus, inrecognition process, the cluster is identified first and then the actual character.

Hidden Markov Modeling

A hidden markov model(HMM) is a statistical model in which the systembeing modeled is assumed to be a Markov process with unobserved state. TheHidden Markov Model is a finite set of states, each of which is associated witha probability distribution. Transitions among the states are governed by a setof probabilities called transition probabilities. In a particular state an outcomeor observation can be generated, according to the associated probabilitydistribution. It is only the outcome, not the state visible to an external observerand therefore states are "hidden" to the outside; hence the name Hidden Markov Model .A HMM can be represented by a Finite State Machine, which in turn can be represented by either a connected graph or a special form of connected graphcalled a trellis. Each node in this graph represents a state, where the signalbeing modeled has a distinct set of properties and each edge a possibletransition between two states at consecutive discrete time intervals. An example of a trellis and graph of a 4 state fully connected HMM is shown in figure below .

Mathematically Hidden Markov Model contains five elements.

I. Internal States: These states are hidden and give the flexibility tomodel different applications. Although they are hidden, usually there issome kind of relation between the physical significance to hiddenstates.

II. Output: O = {O1, O2, O3, . . ., On} an output observation alphabet.

III. Transition Probability Distribution: A = aij is a matrix. The matrixdefines what the probability to transition from one state to another is.

IV. Output Observation: Probability Distribution B = bi (k) is probabilityof generating observation symbol O(k) while entering to state *i* isentered.

V. The initial state distribution (π = {πi}) is the distribution of statesbefore jumping into any state.

Here all three symbols represents probability distributions i.e. A, B and π. The

probability distributions A, B and π are usually written in HMM as a compact

form denoted by lambda as λ = (A, B, π) [17].

Application of OCR across Industry

Optical Character Recognition has a wide range of application across industry verticals which have been gaining traction since the beginning of OCR'sinception. Lately, with the exponential advances in imaging technology, computational processing and Big Data technologies, applicability and adoption of OCR has improved significantly.

1. Insurance

a. Fraud Mitigation: Leverage OCR for extraction of digital information from physical records (legacy and current) and assesses relevant part to deduce any anomalies in the contract vs. claims vs. claim conditions vs. scenarios.

b. Risk Adjustments: Identify the risks associated with the insurance (health or personal) through digital extraction of data from claims documents and identify conditions associated with certain high-medium-low risk profiles to optimally assess the premium.

2. Healthcare

a. Patient Record Digitization: Most patients'records are in physical format,which contains rich information about patient past history, medical conditions and inhibitions, which when digitized and recorded as structured data can be analyzed for downstream real-time medical assessment, re-admission analysis and treatment anomalies.

b. Disease Diagnosis and patient-at-risk: OCR with CAD (Computer-aided Diagnosis) can help identify critical medical conditions from digital records and assist identification of pre-condition(s).

3. Banking

a. Customer Record Digitization: Offline surveys, contract forms and customer documents in digital format contain rich information which, when codified, adds to customer 360 profile and help assess customer information in real-time across sources.

4. Manufacturing / Telecommunications

a. Vendor Contract Assessment: Offline surveys, contract forms and vendor documents in digital format contain rich information which, when codified, adds to vendor 360 profile and help assess customer information in real-time across sources

5. Public / eGovernance

a. Public Record Digitization: Most government records are either in paper or digital form of paper records which, when codified using OCR, can transform eGovernance initiatives.

6. Road Safety

a. Interstate Vehicle Record: Digitization and codification of vehicle records across states can help reduce thefts, criminal activities and terrorism (using stolen vehicles), which can be identified usingcodified records.

b. Toll Identification: Helps identify the type and volume of cars for differential tolls.

Conclusion

Optical Character Recognition (OCR) is becoming a key enabler as companies invest in electronic records. Acknowledging the value of legacy data, OCR is the stepping-stone toward text-based data analytics. Thanks to all those members who experience in Computer Vision, Pattern Recognition and Artificial Intelligence, many teamshas developed the infrastructure necessary to support OCR on a large scale for its clients. Whether the digital conversion project is task-specific or enterprise-wide, OCR infrastructure can be tailored to enable client organizations gain from full-text analytics based on their legacy documents.

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A STUDY ON COGNITIVE SYSTEMS, CAPABILITIES AND THEIR APPLICATIONS

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Abstract: It's not surprising that thepublic's imagination hasbeen ignited by Artificial Intelligence since the term was first coined in 1955.In the ensuing 60 years, we have been alternately captivated by its promise, wary of its potential for abuse and frustrated byits slow development.

Cognitive computing refers to systems that learnat scale, reason with purpose and interact withhumans naturally. Rather than being explicitlyprogrammed, they learn and reason from theirinteractions with us and from their experiences withtheir environment. They are made possible byadvances in a number of scientific fields over thepast half-century, and are different from the information systems that preceded them.Those systems have been deterministic; cognitivesystems are probabilistic. They generate not justanswers to numerical problems, but hypotheses,reasoned arguments and recommendations aboutmore complex — and meaningful — bodies of data.

What's more, cognitive systems can make senseof the 80 percent of the world's data that computerscientists call "unstructured." This enables themto keep pace with the volume, complexity andUnpredictability of information and systems in the modern world.This paper explains thehistory of computing and the rise in cognitive world.

Keywords:Artificial Intelligence,Cognitive Computing.

INTRODUCTION

Cognitive computing (CC) describes technology platforms that, broadly speaking, are based on the scientific disciplines of Artificial Intelligence and Signal Processing. These platforms encompass machine learning, reasoning, natural language processing, speech and vision, human-computer interaction, dialog and narrative generation and more.A wide range of international efforts has been focused on the studies of the new generation of intelligent computers known as cognitive computers, which also known as intelligent computers, brain-like computers, artificial brains, and human centric computers in related research. A *Cognitive Computer* (CC) is an intelligent computer for knowledge processing as that of a conventional von Neumann computer for data processing.

The development of cC is centric in cognitive computing research. *Cognitive Computing* (CC) is an emerging paradigm of intelligent computing methodologies and systems based on cognitive informatics that implements computational intelligence by autonomous inferences and perceptions mimicking the mechanisms of the brain.

To understand the future of cognitive computing, it's important to place it in historical context. To date,there have been three distinct eras of computing — the Tabulating Era,the Programming Era and the Cognitive era.

The Tabulating Era (1900s — 1940s)

The birth of computing consisted of single-purposemechanical systems that counted, using punchedcards to input and store data, and to eventuallyinstruct the machine what to do (albeit in a primitiveway). These tabulation machines were essentiallycalculators that supported the scaling of both businessand society, helping us to organize, understand, andmanage everything from population growth to theadvancement of a global economy.

The Programming Era (1950s — present)

The shift from mechanical tabulators to electronicsystems began during World War II, driven bymilitary and scientific needs. Following the war,digital "computers" evolved rapidly and moved intobusinesses and governments. They performedif/then logical operations and loops, with instructionscoded in software. Originally built around vacuumtubes, they were given a huge boost by the inventionof the transistor and the microprocessor, which came to demonstrate "Moore's Law," doubling incapacity and speed every 18 months for sixdecades. Everything we now know as a computingdevice from the mainframe to the personalcomputer, to the smartphone and tablet is aprogrammable computer.

The Cognitive Era (2011 —)

The potential for something beyond programmablesystems was foreseen as far back as 1960, whencomputing pioneer J.C.R. Licklider wrote hisseminal paper "Man-Computer Symbiosis." Muchof modern computing is based on Licklider'sresearch and insights:

"M an-computer symbiosis is an expecteddevelopment in cooperative interaction between men and electronic computers. It will involve every close coupling between the human and the electronic members of the partnership".

The mainaims are:

1. To let computers facilitate formulative thinking as they now facilitate the solution of formulatedproblems, and

2. To enable men and computers to cooperatein making decisions and controlling complex

situations without inflexible dependence onpredetermined programs…

Preliminary analyses indicate that the symbioticpartnership will perform intellectual operations muchmore effectively than man alone can perform them."

Licklider knew that cognitive computing would bea necessary and natural evolution of programmablecomputing, even if he didn't yet know how it wouldbe accomplished. Fifty years later, massivelyparallel computing and the accumulation of oceansof structured and unstructured data would lay thegroundwork for cognitive computing.

The World's firstCognitive system

In February 2011, the world was introduced toWatson, IBM's cognitive computing system that

Defeated Ken Jennings and Brad Rutter at Jeopardy!. It was the first widely seen demonstration ofCognitivecomputing and it marked the end of theso-called AI winter. The programmable systemsthat had revolutionized life over the previous sixdecades could not have made sense of the messy,Unstructured data required to play Jeopardy!. Watson's ability to answer subtle, complex,pun-laden questions made clear that a new eraof computing was at hand.The true potential of the Cognitive Era will berealized by combining the data analytics andstatistical reasoning of machines with uniquelyHuman qualities, such as self-directed goals,common sense and ethical values. "Teams of human plus machine dominated even thestrongest computers. Human strategic guidancecombined with the tactical acuity of a computer wasoverwhelming. We [people] could concentrate onstrategic planning instead of spending so much timeon calculations. Human creativity was even moreparamount under these conditions.

The technical path forward and the scienceof what's possible

In a global economy and society where valueincreasingly comes from information, knowledge and services, this data represents the mostabundant, valuable and complex raw material in the world. And until now, we have not had themeans to mine it effectively.Programmable systems are based on rules thatshepherd data through a series of predeterminedprocesses to arrive at outcomes. While they arepowerful and complex, they are deterministic — thrivingon structured data, but incapable ofprocessing qualitative or unpredictable input.This rigidity limits their usefulness in addressingmany aspects of a complex, emergent world,where ambiguity and uncertainty abound.

Cognitive systems are probabilistic, meaningthey are designed to adapt and make sense of the

complexity and unpredictability of unstructuredinformation. They can “read” text, “see” imagesand “hear” natural speech. And they interpret thatinformation, organize it and offer explanations ofwhat it means, along with the rationale for theirconclusions. They do not offer definitive answers.

In fact, they do not “know” the answer. Rather,they are designed to weigh information and ideas from multiple sources, to reason, and then offerhypotheses for consideration. A cognitive system assigns a confidence level to each potential insightor answer.However, cognitive systems can learn from their mistakes.Large-scale machine learningis the process by whichcognitive systems improvewith training and use.Many products and servicesthat we use every day — fromsearch-engine advertisingapplications to facial recognitionon social media sites to“smart” cars, phones andelectric grids — are beginningto demonstrate aspects ofArtificial Intelligence.Most consist of purpose-built, narrowly focusedapplications, specific to a particular service. Theyuse a few of the core capabilities of cognitivecomputing. Some use text mining. Others use imagerecognition with machine learning. Most are limitedto the application for which they were conceived.

Cognitive systems, in contrast, combine fivecore capabilities:

1. They create deeper human engagement:

Cognitive systems create more fully humaninteractions with people — based on the mode, form and quality each person prefers. They takeadvantage of the data available today to create a fine-grained picture of individuals — such asgeolocation data, web interactions, transactionmedical records and data from wearable’s —and add to that picture details that have beendifficult or impossible to detect: tone, sentiment,emotional state, environmental conditions and thestrength and nature of a person’s relationships.They reason through the sum total of all thisstructured and unstructured data to find what reallymatters in engaging a person. By continuouslylearning, these engagements deliver greaterand greater value, and become more natural,anticipatory and emotionally appropriate.

2. They scale and elevate expertise:

Every industry’s and profession’s knowledge isexpanding at a rate faster than any professional can keep up with — journals, new protocols, newlegislation, new practices and entirely new fields.A clear example is found in healthcare, where it is estimated that in 1950, it took 50 years todouble the world’s medical knowledge; by 1980,seven years; and in 2015, less than three years.Meanwhile, each person will generate one milliongigabytes of health-related data in his or herlifetime, the equivalent of about 300 million books.

3. They infuse products and serviceswith cognition:

Cognition enables new classes of products andservices to sense, reason and learn about theirusers and the world around them. This allows forcontinuous improvement and adaptation, and foraugmentation of their capabilities to deliver usesnot previously imagined. We see this happeningalready with cars, medical devices, appliances andeven toys. The Internet of Things is dramaticallyexpanding the universe of digital products andservices — and where code and data go,cognition can now follow.

4. They enable cognitive processesand operations:

Cognition also transforms how a companyoperates. Business processes infused withcognitive capabilities capitalize on thephenomenon of data, from internal and externalsources. This gives them heightened awarenessof workflows, context and environment, leadingto continuous learning, better forecasting andincreased operational effectiveness — along withdecision-making at the speed of today’s data.This is good news in a world where, for example,an average billion-dollar company spendsalmost 1,000 person hours per week managingits suppliers.

5. They enhance exploration and discovery:

Ultimately, the most powerful tool that cognitivebusinesses will possess is far better “headlights”into an increasingly volatile and complex future.Such headlights are becoming more importantas leaders in all industries are compelled to placebig bets — on drug development, on complexfinancial modeling, on materials science innovation,on launching a startup. By applying cognitivetechnologies to vast amounts of data, leaderscan uncover patterns, opportunities andactionable hypotheses that would be virtuallyimpossible to discover using traditional researchor programmable systems alone.

Implications andobligations forthe advance ofcognitive science

The Cognitive Era is the next step in the applicationof science to understand nature and improve thehuman condition. In that sense, it is a new chapter ofa familiar story, and the controversy surroundingArtificial Intelligence is merely the latest example ofthe age-old debate between those who believe inprogress and those who fear it. Within the scientificcommunity — as opposed to the media and popularentertainment — the verdict is in. There is broadagreement on the importance of pursuing a cognitivefuture, along with recognition of the need to developthe technology responsibly. “Technology creates possibilities and potential,but ultimately, the future we get will depend on the choices we make. Technology is not destiny.We shape our destiny.” Specifically, we must continue to shape the effect ofcognitive computing on work and employment. Likeall technology, cognitive computing will change thenature of work done by people. It will help us performsome tasks faster and more accurately. It will makemany processes cheaper and more efficient.

Some features that cognitive systems may express are:

- **Adaptive**: They may learn as information changes, and as goals and requirements evolve. They may resolve ambiguity and tolerate unpredictability. They may be engineered to feed on dynamic data in real time, or near real time.[13]
- **Interactive**: They may interact easily with users so that those users can define their needs comfortably. They may also interact with other processors, devices, and Cloud services, as well as with people.
- **Iterative and stateful**: They may aid in defining a problem by asking questions or finding additional source input if a problem statement is ambiguous or incomplete. They may “remember” previous interactions in a process and return information that is suitable for the specific application at that point in time.
- **Contextual**: They may understand, identify, and extract contextual elements such as meaning, syntax, time, location, appropriate domain, regulations, user’s profile, process, task and goal. They may draw on multiple sources of information, including both structured and unstructured digital information, as well as sensory inputs (visual, gestural, auditory, or sensor-provided).

Some important Applications of Cognitive Systems are:

1. Human Face Recognition from2D to 3D

Human beings are born with a natural capacity of recovering shapes from merely a single image. However, it remains a significant challenge in cognitive computing and AI to let a computer obtain such ability.

2.High-Frequency Cognitive Processes

Highfrequency cognitive processes are currently perceived as beneficial to critical decision making and ultimately to a higher predictive power on immediate future and the impact of events in our society.

3.Intelligent Pattern Recognition and Applications

Intelligent Pattern Recognition (IPR) and applications include the following: 3D biometric technology and applications in social and national security.

4. Granular Computing and Conceptual Modeling of Cognitive Computing

Conceptual modeling is an important perspective on cognitive informatics and cognitive computing . Cognitive computing involves extremely complicated processes because its carrier is the brain. A suitable conceptual model enables us to focus on the main features at a more abstract level, without being overloaded with minute details and without worrying about physical or biological implementations.

5.Artificial Brains and Cognitive Computing

Sufficient progress has been made with Moore’s Law and with knowledge of the microcircuitry of the human brain. Time is now ripe for large scale artificial brains to be built that mimic closely the scale and detailed structure of the human brain.

CONCLUSION

In the end, all technologyrevolutions are propelled notjust by discovery, but alsoby business and societalneed. We pursue these newpossibilities not because wecan, but because we must.“The greatest bar to wise action and the greatestsource of fear is ignorance. A tiny candle givesmisleading light and throws huge and ominousshadows. The sun at noon gives great lightand throws no shadows. It is time to get this wholeproblem of men and machines under a blazing noonday beam. Computers will never rob manof his initiative or replace the need for his creativethinking. By freeing man from the more menial or repetitive forms of thinking, computers willactually increase the opportunities for the full useof human reason."In so doing, we will pave the way for the nextgeneration of human cognition, in which we thinkand reason in new and powerful ways. It's true thatcognitive systems are machines that are inspiredby the human brain. But it's also true that thesemachines will inspire the human brain, increase ourcapacity for reason and rewire the ways in whichwe learn. In the 21st century, knowing all the answerswon't distinguish someone's intelligence — rather,the ability to ask better questions will be the markof true genius.

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SEQUENTIAL PROBABILITY TREE FOR MINING USER COMMUNITIES FROM GEO-CLUSTER

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Abstract – Sequential Pattern mining is a topic of data mining concerned with finding statistically relevant patterns between data examples where the values are delivered in a sequence. To address the Problem of mining movement based communities from user's trajectories to propose a frame work to mine movement based communities include to construct sequential probability tree (SP-Tree) in a level by level manner by using breadth first approach and depth first Approach of algorithms and capture sequential patterns and indicate transition probabilities of movements, to calculate similarity functions include two existing common similarity function and one new similarity function that explores all of the information of the tree structures, In light of the similarity values derived to formulate an objective function to evaluate the quality of communities to propose a algorithm Geo-cluster to effectively desire communities.

Keywords- Trajectory Pattern Mining, Clustering, Sequential patterns, Trajectory Profile, Community Structure.

1. INTRODUCTION

Trajectories are sequences that contain the spatial and temporal information about movements. Trajectory pattern mining algorithm is genuinely spatio-temporal, in the sense that the identification of the regions of interest is dynamically intertwined with the mining of sequences with temporal information. This approach is capable to detect more precise trajectory patterns, as the regions of interest are incrementally identified as locally dense regions, i.e., with respect to the trajectories in the patterns found so far. The development of positioning techniques, many portable devices are equipped with location aware sensors, meaning that the positions of uses can be obtained some movement based community application examples are listed below.

1. Trajectory ranking:

Many of the websites provide a query interface, depicting some query predicative Requirements, to retrieve user **trajectories.**

2. Community based traffic sharing services:

For example users can download the mobile application, waze, to share traffic information and use the navigation service. With the development of positioning techniques, many portable devices (e.g., smart phones) are equipped with location-aware sensors, meaning that the positions of Users can be easily obtained. Some application examples are Trajectory ranking, Community-based traffic sharing services, Friend recommendation**.** With the movement-based community, one could recommend possible users who are likely to have the same or similar movement preferences. For example, by recommending some users who have similar running behaviors, a group-based running activity could be easily initiated.To discover movement-based communities, one should first formulate the similarity of users in terms of their trajectories [1]. Clearly, since trajectories are time series data, one naive idea is to explore existing similarity functions, such as Euclidean distance, DTW, EDR, ERP and LCSS [2], to mea-sure the similarity degree of users. However, since trajectories contain uncertainty and raw noise data, it is hard to accurately model the movement similarity of users via using existing similarity functions from user raw trajectories. Moreover, the computing cost is expensive if one would like to derive the similarity value among users from their raw trajectory data. To bridge the uncertainty and noise in trajectory data, without loss of generality, most prior works [6], [9] extract regions, where one user frequently visits. Those regions are likely to represent hot regions or popular regions from trajectories. Then, prior works [6], [9] explore the sequential relationships among the extracted regions. These sequential regions are able to be viewed as the determinable features to describe movement behavior. As in [5], [10], we utilize a grid-based approach to extract hot regions. As such, each trajectory could be represented by a sequence of hot regions (referred to as a transformed trajectory).

To capture movement behavior from a set of transformed trajectories, one could use sequential pattern mining methods to extract frequent sequential patterns. The frequent sequential patterns are the subsequences of the transformed trajectories if the number of transformed trajectories containing the subsequences is larger than a pre-defined thresh-old (i.e., the minimum support). Further-more, the conventional sequential patterns are derived by the frequency of the sequences which occur in the trajectories and do not have transition probabilities among hot regions. Therefore, to compare the movement behaviors effectively, considering both the movement sequences and probability of mobility transition is necessary.

We first propose the concept of trajectory profiles to characterize users' movement behaviors, where the trajectory profile is a tree structure with frequent movement sequences and transition probabilities. Then, based on the free-based trajectory profile, we propose a new similarity function to model user similarity in terms of their trajectory profiles. Furthermore, we formulate the objective function of clustering and propose one algorithm to cluster users into groups. Note that each group represents one movement based community.In our proposed framework, several technical issues are addressed:

Constructing user trajectory profiles: A trajectory profile should capture two kinds of movement information: a) the sequential patterns of hot regions, and b) the transition probabilities among hot regions. The sequential patterns among hot regions describe the ways in which trajectories frequently transit among these regions. Here, we propose a compact data structure, called Sequential Probability Tree (SP-tree), to mine both sequential patterns and transition probabilities. Explicitly, in an SP-tree, each node is labeled by tree edges traversed from the root node of the SP-tree, and labels represent sequential patterns. For each node, a conditional table that depicts the next movement from a tree node, is constructed. Instead of capturing all transition probabilities among hot regions, the conditional table of nodes only reflects the next movement probabilities. By traversing SP-trees, one could derive transition probabilities of hot regions. Furthermore, the proposed SP-tree is a data structure to compress the sequential patterns into a compact tree model, which indicates the sequential patterns and the transition probabilities.

Deriving similarity between trajectory profiles:

To distinguish how close two trajectory profiles are, we need to develop a similarity function between two SP-trees. Since trajectory profiles are tree structures, we formulate some similarity functions, including two existing common similarity functions and one new similarity function that explores all of the information of the tree structures. Four details of the tree structures, such as the number of common tree nodes, the number of different tree nodes, the support values of the tree nodes and the conditional tables of the tree nodes, are considered in the proposed SP-tree based similarity function.

Discovering movement-based communities:

Given a set of users with their similarity scores, a geo-connection graph is built for discovering movement-based communities. Each node of the geo-connection graph represents each user, and the weight edge depicts the similarity score between two users. A geo-connection can be constructed among users if their similarity scores are larger than a pre-defined thresh-old that could be viewed as a minimum similarity bound in movement-based communities. Given a geo-connection graph, our goal is to cluster vertices into groups, where each group represents one community. Thus, we formulate an objective function to measure the quality of the cluster results. In light of the objective function, we propose algorithm Geo-Cluster to cluster users in the geo-connection graph. In addition, since the number of user communities is usually not known in advance, the number of groups should not be assigned when partitioning users into communities. In this paper, we reveal a way to mine user communities without specifying the number of communities.

2. RELATED WORK

The goal is to discover user communities based on user movement behavior hidden in user trajectories. Our work is related to trajectory pattern mining in that we intend to mine user movement behavior from trajectories. Thus, we will present some existing works on trajectory pattern mining.

2.1 Trajectory Pattern Mining

A considerable amount of research effort has already been put into mining movement behavior from trajectories. Given a set of trajectories, trajectory pattern mining aims to discover frequent sequential patterns that are sequential relationships among regions. Generally speaking, the flow of mining trajectory patterns is to find hot regions, and then discover sequential relationships among them. The authors in [8] proposed a clustering-based approach to discover moving regions within time intervals. In [5] and [3], the authors exploited temporal annotated sequences in which the sequences are associated with time information (i.e., transition times between two movements). In this paper, we intend not only to discover user movement behavior but also to utilize movement behavior to discover movement-based communities. Furthermore, we cluster users into different communities according to user trajectories. The work in [7] is to cluster trajectories with a similar shape into spatial and temporal domains, which targets clustering trajectories and not users as addressed in our proposed work.

2.2 User Movement-Based Community Mining

In [9], [13], the authors aim to find user similarity based on their trajectories. However, the proposed work in [9], [13] uses the sequential patterns to format user similarity. They extract similar sequences in two users' trajectories, and then calculate the similarity of each. There is a higher similarity score if there are many similar location sequences and similar time durations between two locations among their trajectories. After deriving the similarity of each similar sequence, the similarity between two users is the sum of all the similarities of each similar sequence in their trajectories. In [28], the authors aim to reduce the transmission cost in sensor networks. They propose a clustering method to aggregate objects with similar moving traces. They use PST to represent the movement profile of moving objects. To cluster the moving objects based on the PST, they calculate the difference of probability of all possible path in two PSTs, and then normalize and convert the value into similarity. Based on the similarity measurement, a distributed clustering method is proposed to cluster similar moving objects. Given a set of trajectories, the goal is to cluster trajectories into groups, where trajectories in the same group have a certain degree of similarity. The primary goal of this work is to discover common sub-trajectories from a set of trajectories. However, in this paper, the goal is to cluster users into groups according to their own set of trajectories. Therefore, the existing works of trajectory clustering cannot be directly applied to model user similarity.

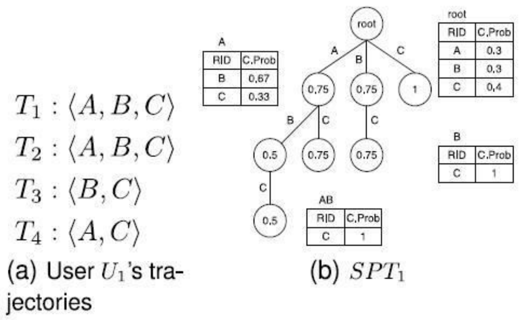


Fig. 1. User 1's trajectories and SP-tree SPT_1.

Given a set of trajectories of user U1 in Fig. 1a, Fig. 1b is the corresponding SP-tree for user U1. As can be seen in Fig. 1b, the SP-tree contains a root node and a set of tree nodes labeled as {A, B, C, AB, AC, BC, and ABC}. As can be seen in Fig. 1b, the SP-tree contains a root node and a set of tree nodes labeled as {A, B, C, AB, AC, BC, and ABC}. The node along with the leftmost branch is root -> A -> AB -> ABC, which represents the frequent sequential patterns (i.e., A, AB, and ABC). If the support threshold is set to 2, one could verify that A, AB, and ABC are indeed the frequent subsequences with their supports equal to 2. Moreover, the conditional table associated with each node represents the probabilities of the next possible movements of the sequential pattern of this node. For example, the conditional table of node A contains two entries: (B, 0.67) and (C, 0.33). This means that the user goes from the hot region A to the hot region B with a probability of 0.67 and to the hot region C with a probability of 0.33.

Note that SP-tree is able to not only capture sequential movement patterns but also indicate transition probabilities among hidden hot regions. Furthermore, SP-Tree is a com-pact structure in that nodes in the upper levels are shared with nodes in the lower levels. Usually, sequential patterns may have the same or similar prefix sequences. These prefix sequences will be represented as nodes in the upper levels of SP-trees, which is in fact a compact data structure. Since SP-tree should capture frequent user movement behavior, two minimal threshold values: MinSup and MinProb are used to guarantee that the support of each node should be larger or equal to MinSup, and the corresponding probability of the appearing count for each tree node should be larger than MinProb.

3. ALGORITHMS FOR CONSTRUCTING SP-TREES

Given a set of one user trajectory and two thresholds (i.e., MinSup and MinProb), in this section, we propose two algo-rithms to construct an SP-tree as a user trajectory profile. The first algorithm explores a breadth first idea, called algorithm BF, to derive SP-trees in a level by level manner. On the other hand, the other algorithm, named DF, exploits the depth-first concept in deriving SP-trees. In addition, both time and space complexity analysis of algorithms BF and DF are presented.

3.1 Algorithm BF: A Breadth-First Approach

Algorithm BF is proposed to construct SP-trees in a level-by-level manner. Initially, SP-tree has only one root node with the conditional table which has the probabilities of hot regions being larger than a minimum probability threshold MinProb.

When the support value of a hot region is larger than the minimal support threshold MinSup, a node is created in the second level. In each level, find frequent hot regions (line 7) and construct the conditional table of each node (line 8). Then, determine child nodes of each node (line 9 to line 13).Fig. 2 shows the process of constructing the SP-tree of user U1, where the settings are MinSup = 0.4 and MinProb =0.3. In the beginning, the root node represents the empty sequential pattern. Thus, S0 ={root}. Then, calculating the frequent locations and conditional table of each node in S0, there is only one node, root, in S0. The frequent hot regions of the projected trajectory set of root are A, B and C since their support value are 3/4, 3/4 and 4/4, respectively. Particularly, the projected trajectory set of root is T .

Algorithm 1 BF

Input: A set of transformed trajectories $\mathcal{T}$, a minimum probability threshold $MinProb$, a minimum support threshold $MinSup$
Output: An SP-tree SPT
1: SPT = a root node with empty entry;
2: $S_0 = \{root\}$;
3: $k = 0$;
4: **while** $S_k \neq \emptyset$ **do**
5: $S_{k+1} = \emptyset$
6: **for each** node s **in** S_k **do**
7: find frequent hot regions
8: create conditional table of node s
9: **for each** σ **in** frequent hot regions **do**
10: **if** σ in conditional table of node s **then**
11: create project transformed trajectory set of $s\sigma$
12: $s\sigma$ is a child of s
13: add node $s\sigma$ into S_{k+1}
14: **end if**
15: **end for**
16: **end for**
17: $k = k + 1$;
18: **end while**

As for the conditional table of root, there are three hot regions (i.e., A, B, and C) in U1's trajectories. As can be seen in Fig. 2, the number of regions in U1's trajectories is 10. Among these trajectories, A appears three times, B appears three times, and C appears four times. Thus, the C:Prob of A, B, and C in the conditional table of root are 3/10, 3/10, and 4/10, respectively. Because the C:Prob of all entries in the conditional table of root are larger than MinProb, the conditional table of root contains three entries (Fig. 2a).

Then, we could derive the child nodes of root. For each frequent hot region, we inspect whether it is in the conditional table of root or not. Since all frequent hot regions are in the conditional table of root, root has three child nodes, and S1={A;,B, C} (Fig. 2b). From S1, we could further have S2. As such, the frequent hot regions of node A are B and C, where the projected trajectory data set is {(B,C),(B,C)(B,C)}. There are two entries, B and C, in the conditional table of node A. Thus, S2 = {AB, AC}, After the same procedure for nodes B and C, S2 ={AB, AC, BC} (Fig. 2c). After finding frequent hot regions and creating the conditional table, we create child nodes of the nodes in S1 (Fig. 2d). Following the above procedure, the final SP-tree is shown in Fig. 1b.

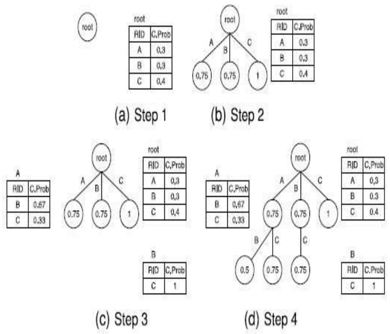


Fig. 2. An example of constructing SP-tree of user U_1 using algorithm BF.

3.2 Algorithm DF: A Depth-First Approach

Note that algorithm BF constructs the SP-tree level by level. To create child nodes for the next level, algorithm BF needs to have a buffer Sk to maintain processing nodes. Then, it iteratively selects one tree node from the buffer and deter-mines possible candidate child nodes from the selected tree node until the buffer is empty. Those candidate child nodes still need to put in the buffer to explore more child nodes. Consequently, algorithm BF is likely to degrade its performance if the SP-tree is flat.

Algorithm 2 DF

Input: A set of transformed trajectories $\mathcal{T}$, parent node s, a minimum probability threshold $MinProb$, a minimum support threshold $MinSup$
Output: An SP-tree SPT
1: find frequent locations
2: create conditional table of node s
3: **for each** σ **in** frequent locations **do**
4: **if** σ in conditional table of node s **then**
5: $s\sigma$ is a child of s
6: create projected transformed trajectory set $\mathcal{T}'$
7: DF($\mathcal{T}'$, $s\sigma$, $MinProb$, $MinSup$)
8: **end if**
9: **end for**

The reason is that a considerable amount of memory space is for storing the projected trajectory set of a flat SP-tree. As such, we further propose algorithm DF, which is a depth-first approach, to derive the SP-trees. In the beginning, algorithm DF generates one root node in the SP-tree and its conditional table which has the probability of hot regions being larger than a minimum probability threshold MinProb (line 1 to line 2). For each frequent hot region, algorithm DF will check whether it is in the conditional table or not. If it is in the conditional table with its probability larger than MinProb, algorithm DF will generate one tree edge labeled as the corresponding frequent hot region and one child tree node adjunct with the tree edge. For the new child tree node, algorithm DF will further derive the conditional table of the new child node. It will then further identify possible frequent hot regions from the conditional table. Note that algorithm DF is recursively performed (line 5 to line 7). Hence, a depth-first trace from the root node to the leaf tree node is derived.

For example, to construct SP-tree SPT1 for user U1, the setting of MinSup and MinProb is the same as the setting in 4.1. Fig. 3 shows the process of constructing an SP-tree using algorithm DF. In the beginning, DF(T , root, 0.3, 0.4) is called, and there is only one root node in SPT1. Then, the frequent locations of transformed trajectories$T^1$are A, B, and C (line 1), and there are three entries A, B, and C in the conditional table of root (line 2, Fig. 3a). To create child nodes of root, the procedure checks whether each frequent location is in the conditional table (line 4). In this case, frequent locations, A, B, and C, are all in the conditional table. A is a child of root (line 5). Thus, DF(T^1 A, 0.3, 0.4) is called in line 7, where T^{0} is {{B,C}{B,C},{C}}(Fig. 3b). Following the procedure, DF(T^{1}, AB, 0.3, 0.4) calls DF(T^{1}, ABC, 0.3, 0.4) (Fig. 3c), and DF(T^{1}, A, 0.3, 0.4) calls DF(T^{1}, AC, 0.3,0.4) after calling DF(T^{1}, AB, 0.3, 0.4) in the for each loop (line 3). The final SP-tree is shown in Fig. 1b.

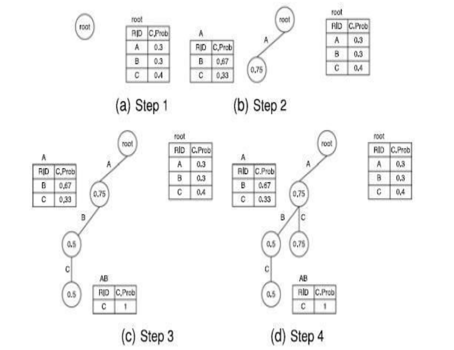


Fig. 3. An example of constructing the SP-tree of user U_1 using algorithm DF.

4. MOVEMENT-BASED COMMUNITY DISCOVERY

In this section, we present how to cluster users in terms of their trajectory profiles (i.e., SP-trees). Given a set of users with their SP-trees, our goal is to cluster users into groups where users in the same group have a certain degree of movement similarity. To achieve this goal, similarity functions based on SP-trees are presented. In light of the devised similarity degrees, we further propose one clustering algorithm to derive communities of users. The following sections show our approaches for calculating similarity and discovering communities in detail.

4.1 Similarity Function Based on SP-Tree Structures

As pointed out earlier, SP-trees of users capture user movement behavior. We thus design a new similarity function that explores the tree structures of SP-trees. Because an SP-tree contains both sequential patterns and transition probabilities of hot regions, the design of our proposed similarity function, denoted as Simsp, will consider the following tree structure information:

1. The number of common tree nodes. A node in an SP-tree is in fact a sequential pattern, which indicates a frequent sequence of hot regions from user trajectories. As such, the more common nodes that two SP-trees have, the more similar the movement behavior of the two users. For example, consider SP-trees SPT1, SPT2 and SPT4 in Fig. 1b and Fig. 4. The number of common nodes between SPT1 and SPT2 is 5 (i.e., root, A, B, C, and AC), while between SPT1 and SPT4 is 4 (i.e., root, B, C, and BC). Clearly, SPT1 is more similar to SPT2 than to SPT4.

2. The number of different tree nodes. Due to the fact that the number of nodes in an SP-tree captures frequent user movement regions, the similarity function should consider the different number of nodes in SP-trees. The reason is that it is possible that two users may have common nodes in their SP-trees but their movement coverage ranges are different. For example, user Ui may have a variety of movement ranges and have an SP-tree with more number of nodes. On the other hand, if user Uj does not travel a lot and thus has a smaller movement coverage range and if these two users have some common nodes in their SP-trees, they are likely to have similar movement behavior. However, these two users have different movement behavior in terms of their movement coverage ranges. Thus, the number of different nodes in the SP-trees should be considered in the similarity function.

3. **Support values of tree nodes.** The support value of a node indicates how frequently the sequential pattern appears among user trajectories. Different support values of sequential patterns represent different movement behavior. Clearly, the closer the support values of two common nodes are, the more similar their movement behavior is. For example, SPT3 and SPT4 have the common nodes B and BC with the same support values (i.e., 1 and 0.5). In the case of SPT1 and SPT3, their common nodes B and BC have different support values. Consequently, the sequential

patterns B and BC between SPT3 and SPT4 are more similar than those between SPT1 and SPT3. Furthermore, the support value of a sequential pattern also refers to the appearing frequency of this pattern. Obviously, the higher the support value is, the more important the sequential pattern is. Therefore, the support values of nodes in SP-trees should infer the importance of nodes in deriving the similarity scores.

4. **Conditional tables of tree nodes**. Each node in an SPtree has the conditional table that demonstrates the next movements. From the conditional tables of the tree nodes, one could have the transition probability of the hot regions, which indicate more detailed movement behavior of users. Note that it is possible that two SP-trees have the same sequential pattern but the next movement probabilities are not the same.

Let two SP-trees be SPT_i and SPT_j, and nodes in SPT_i with the sequential pattern s are represented by N_i^s. Given two nodes N_i^s and N_j^s, the similarity score for these two nodes is defined as in Equation (1)

$$Sim_N\left(N_i^s, N_j^s\right) = \begin{cases} 1, \text{ if } s = root \\ \left(1 - \left|sup\left(N_i^s\right) - sup\left(N_j^s\right)\right|\right) \\ \times \dfrac{sup\left(N_i^s\right) + sup\left(N_j^s\right)}{2}, \text{ otherwise.} \end{cases} \quad (1)$$

The conditional table of a node N_i^s is used to record the probabilities of next movements after the sequential pattern s. Clearly, the smaller difference of the two conditional tables of the two common nodes should be more similar in the similarity measurement. Phase 1 only adds the next movement for which the probability exceeds $MinProb$ into a conditional table. However, the sum of next movement probabilities is not equal to 1. In this case, if the sum of probabilities is p, we can still view the probability of a null next movement as $1 - p$. Thus, the conditional table of each node can be viewed as a probability distribution. To evaluate the similarity between two conditional tables, the well-known probability distribution distance can be adopted. Let the conditional table of N_i^s be C_i^s. Then, define the similarity score for the conditional tables using Equation (2)

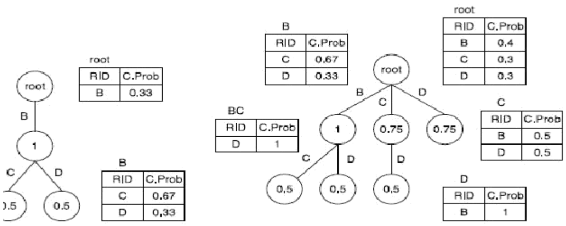
$$Sim_T(C_i^s, C_j^s) = 1 - \frac{\sum_{\rho \in C_i^s \cup C_j^s} |Pr(C_i^s(\rho)) - Pr(C_j^s(\rho))|}{|C_i^s \cup C_j^s|}.$$

Meanwhile, it is much easier to have many different nodes from others as well. Therefore, normalizing the overall similarity based on the total number of nodes in two SP-trees can reflect this phenomenon

$$Sim_{SP}(SPT_i, SPT_j) = \frac{\sum_{s \in S_{i,j}} Sim_N\left(N_i^s, N_j^s\right) \times Sim_T\left(C_i^s, C_j^s\right)}{|SPT_i \cup SPT_j|}.$$

For example, consider SPT1 and SPT2. Although they have the common node A with the same support value 0.75, the conditional table of node A in SPT1 is totally different from that in SPT2. After staying at hot region A, User U1 may go to hot regions B and C. However, User U2 may go to only hot region C. By exploring the probabilities of next movements in the SP-trees, it is more accurate to evaluate how similar two SP-trees are by exploring the conditional tables of the tree nodes in the SP-trees.

4.2 Clustring users based on SP-Tree Similarities



(b) SPT_3 (c) SPT_4

According to the above similarity functions, the similarity scores of users are derived. Then, in light of these scores, we first explore a graph structure to represent users' movement similarity relationships. Based on the graph structure, we formulate an objective function to evaluate the quality of clustering results and thus propose one algorithm to derive clusters.

Two users have a *geo-connection* relationship if the similarity score of their trajectory profiles exceeds a threshold δ. The threshold δ is a pre-defined threshold that could be viewed as a minimum similarity bound in movement-based communities. According to the geo-connection relationships among users, a geo-connection graph is thus built as $G = (V, E)$, where $V = \{v_1, v_2, \ldots, v_n\}$ represents users and $E = \{(v_i, v_j) | Sim_{SP}(SPT_i, SPT_j) > \delta\}$ indicates the similarity scores between user U_i and user U_j. Given a geo-connection graph $G = (V, E)$, our goal is to derive a set of clusters (i.e., sub-graph components) where each sub-graph component represents a set of nodes that have a certain degree of movement similarity.

For a community, use the *intra-cost* score, which refers to the minimum number of edges added into these nodes in this community, to make them a clique. Formally, the intra-cost score of a community $C_i = (V_i, E_i)$ is formulated as in Equation (4)

$$Cost_{intra}(C_i) = |K_{|V_i|}| - |E_i|, \text{ where } K_{|V_i|} \text{ is a } |V_i|\text{-clique.} \quad (4)$$

these nodes are hardly recognized as members of the same community. With communities in a geo-connection graph, the *inter-cost* represents the minimum number of edges removed from the geo-connection graph to make the communities disjointed from each other. Consequently, the inter-cost score of two communities $C_i = (V_i, E_i)$ and $C_j = (V_j, E_j)$ can be formulated as in Equation (5)

$$Cost_{inter}(C_i, C_j) = |\{(v_i, v_j) | v_i \in V_i, v_j \in V_j\}|. \quad (5)$$

To combine these two scores, the objective function for a set of user communities $C = \{C_1, C_2, \ldots, C_n\}$ can be derived as Equation (6)

$$Cost_{total}(C) = \sum_{C_i \in C} Cost_{intra}(C_i) + \sum_{C_i, C_j \in C} Cost_{inter}(C_i, C_j). \quad (6)$$

Based on the derived objective function, algorithm Geo-Cluster, shown in algorithm 3, is proposed. Algorithm Geo-Cluster first constructs a geo-connection graph based on the similarities of trees and the threshold d.

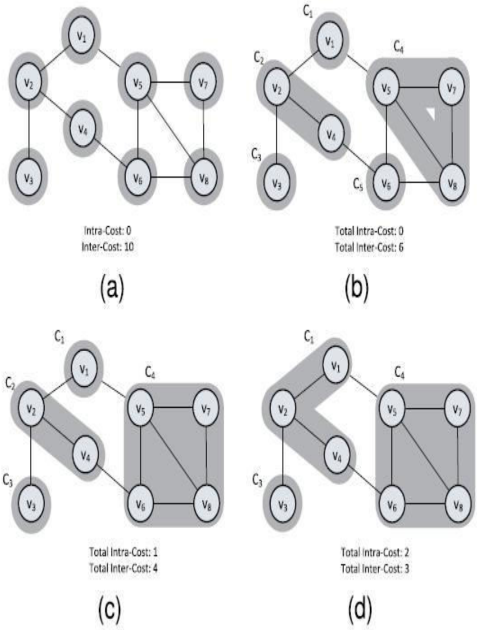


Fig. 5. An example of algorithm Geo-Cluster for mining user communities.

In the beginning, each vertex is viewed as a community (line 2), and the total cost scores for merging any two communities are computed (line 10). Once the total cost can be reduced, two communities are set as the candidate communities (Xi, Xj) in line 13. After exiting the for loop in line 9 to line 15, the two communities that can minimize the total cost score are found, and they are merged into one. This algorithm terminates when the total cost score cannot be further reduced or none of the communities can be expanded.

Algorithm 3 : Geo-Cluster

Input: User profiles: $\{SPT_1, ..., SPT_n\}$, and t
olds: δ

Output: C, communities of users

1: Construct a geo-connection graph $G = (V,$
$\{SPT_1, ..., SPT_n\}$ and δ;
2: $C \leftarrow V$;
3: $LastCost \leftarrow Cost_{total}(C)$;
4: $Flag \leftarrow True$;
5: **while** $Flag = True$ **do**
6: $Flag \leftarrow False$;
7: $(X_i, X_j) \leftarrow (\phi, \phi)$;
8: $MinCost \leftarrow LastCost$
9: **for** each community $C_i, C_j \in C$ **do**
10: $CurrCost \leftarrow LastCost + |C_i \times C_j| - Cost_{inter}(C_i, C_j)$
11: **if** $CurrCost \leq MinCost$ **then**
12: $Flag \leftarrow True$;
13: $(X_i, X_j) \leftarrow (C_i, C_j)$;
14: **end if**
15: **end for**
16: **if** $Flag = True$ **then**
17: Merge X_i and X_j;
18: $LastCost \leftarrow MinCost$;
19: **end if**
20: **end while**

Fig. 5 shows a running example of algorithm Geo-Cluster. Initially, Fig. 5a shows each node as a community. The total intra-cost score is 0 and the total inter-cost score is 10. Fig. 5b shows the results after three merges. Following the result in Fig. 5a, the Last Cost is 10. Geo-Cluster enumerates every pair from C and examines whether the merger of two communities can reduce the Last Cost or not. First, merging v2 and v4 into C2 increases the total intra-cost by 0 and decreases the total inter-score by 1. Therefore, the value of Last Cost decreases by 1. Similarly, both merging v5 and v7 into a community C0 and merging C0 to v8 into a community can reduce the Last Cost since the total intra-cost increases by 0 and the inter-cost increases by 1 and 2, respectively.

After this iteration, there are five communities C1, C2, . . . and C5. Fig. 5c shows an intermediate result in which these five communities have an intra-cost score of 0 and the inter-cost score is 6. Merging C4 and C5 increases the total intra-cost score by 1, while decreasing the total inter-cost by 2. Thus, Fig. 5c shows that C4 and C5 are then merged and the total cost is 5 (i.e., 1 + 4). Fig. 5d shows the final result with three communities C1, C3, and C4.

5. CONCLUSION

We address the problem of mining communities from users' trajectories. We proposed a framework of mining movement-based communities, which consists of three phases: 1) constructing user trajectory profiles, 2) identifying the closeness of users by their profiles, and 3) discovering movement-based communities. In the first phase, for each user, we designed an SP-tree to capture the sequential patterns and the transition probabilities of movements. In the second phase, we formulated some similarity functions, including two existing common similarity functions and one new similarity function that explores all of the information of the tree structures. In the third phase, we formulated an objective function to measure the quality of the cluster results. In light of the objective function, we proposed algorithm Geo-Cluster to cluster users. Due to more information and services being available in location-based social networks, prior works in [5], [2], [12] have proposed mining semantic meanings of regions. In the future, we aim to extract hot regions with semantic meanings. Based on the semantics of hot regions, we will be able to mine communities that have users with similar movement behavior and activity behavior.

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CHALLENGING ISSUES IN CLOUD COMPUTING ALONG WITH BIG DATA MANAGEMENT

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***Abstract*— Cloud computing is a powerful technology to perform massive-scale and complex computing. It eliminates the need to maintain expensive computing hardware, dedicated space, and software. Massive growth in the scale of data or big data generated through cloud computing has been observed. Addressing big data is a challenging and time-demanding task that requires a large computational infrastructure to ensure successful data processing and analysis. The rise of big data in cloud computing is reviewed in this study. The definition, characteristics, and classification of big data along with some discussions on cloud computing are introduced. The relationship between big data and cloud computing, big data storage systems, and Hadoop technology are also discussed. Furthermore, research challenges are investigated, with focus on scalability, availability, data integrity, data transformation, data quality, data heterogeneity, privacy, legal and regulatory issues, and governance. Lastly, open research issues that require substantial research efforts are summarized. This paper introduces several big data processing techniques from system and application aspects here provide an organized picture of challenges that are focused by the application developers and DBMS designers in developing cum deployment of the internet scale applications. Then we see about the security issues in the cloud computing along with the big data and Hadoop. We show some possible solutions for the issues of the cloud computing and Hadoop.**

***Keywords*-Big Data; Cloud Computing;**

I . INTRODUCTION

The successful paradigm for the service oriented programming is the cloud computing. It has revolutionized the way of computing infrastructure's abstraction and usage. The elasticity, pay per use, low upfront investment, transfer of risks are few of the major enabling characteristics that makes the cloud computing the ubiquitous platform for deploying economically feasible enterprise infrastructure settings. Distributed databases had been the boon of vision for research for few decades. But changes in the data patterns and applications has made way for the new type of storage called key value storage which are now being widely used by various enterprises. In the domain of Map reduce [2] and open source implementation of the same known as the Hadoop [3] has been used by majority of the industry and academics. Hadoop increases the usability and performance [4, 5].HDFS has become a Very helping tool to maintain and store the complex data. Big data has becoming more available and understandable to computers. What is big data? The question arrives. Big data is the representation of progress of the human cognitive processes, usually includes data sets with sizes that is beyond the current technology's capability. The data which is very fast, has various varieties and requires new type of the processing forms to enable decision making, insight discovery and optimization of process. In order for analyzing the data and for identification of patterns it is very important for us to store the data securely, manage and sharing of complex data on cloud. Since cloud involves extensive complexity, we feel its ideal to make enhancements in securing cloud than showing holistic solutions.

In this paper we provide a comprehensive background study of state of art systems. Identification of critical aspects in design of various systems and scope of the systems. We show up some approaches in security provision through a scalable system to handle large number of sites and also has the capability to process large and massive amounts of data. We also provide the status of big data studies and related works, aiming at providing a overview of managing big data and its applications.

BIG DATA

Big data is a word used for description of massive amounts of data which are either structured, semi structured or unstructured. The data if it is not able to be handled by the traditional databases and software tech ologies then we categorize such data as big data. The term big data [5] is originated from the web companies who used to handle loosely structured or unstructured data. The big data is defined using three v's.

1) Volume: many factors contribute for the increase in volume like storage of data, live streaming etc.

2) Variety: various types of data is to be supported.

3) Velocity: the speed at which the files are created and processes are carried out refers to the velocity.

Fig 1: Big data

Fig 1 shows a typical bug data representation./ The areas for example that comes in big data are shown.

Technologies not only supports the collections of large amounts such data effectively. Transactions that are made all over the world in a Bank, Walmart customer transactions, and Facebook users generating social interaction data

Are few examples for big data usage.

I. HADOOP

This is a freely available java based programming framework supporting for the processing of large sets of data in a distributed computing environment. Using Hadoop, big amount of data sets can be processed over cluster of servers and apps may be run on system with thousands of nodes involving terabyes of information. This lowers the risk of system failure even when a huge number of nodes fail.it enables a scalable, flexible, fault tolerant computing solution. HDFS[7], a file system spanning all nodes in a Hadoop cluster for data storage links the file systems on local nodes to make it onto a very large file system thus improving the reliability.

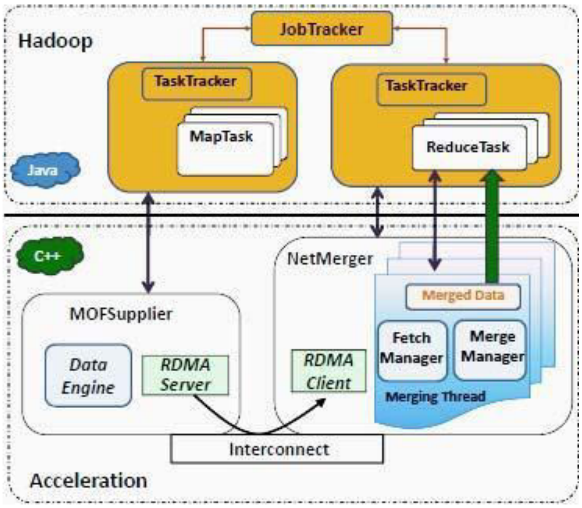

Fig 2: Hadoop structure

Task trackers are responsible for running the tasks that the job tracker assigns them

Job trackers has two primary responsibilities which are managing the cluster resources and scheduling all user jobs

Data engine consists of all the information about the processing the data

Fetch manager helps to fetch the data while particular task is running.

II. MAP REDUCE

Map reduce[8] framework is used to write apps that process a large amounts of data in a reliable and fault tolerant way. The application is initially divided into individual chunks which are processed by individual map jobs in parallel. The output of map sorted by a framework and then sent to the reduce tasks. The monitoring is taken care by the framework.

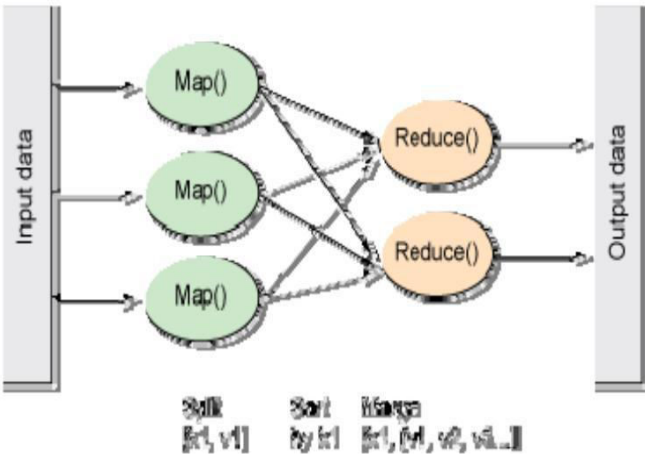

Fig 3: Map reduce

The input data is divided into individual chunks and are provided for processing by the map task. These map task process the data in parallel and the result from the map task is then provided to the reduce task where the results that are generated in parallel by the map task are consolidated and the reduced report is given as output.

Big data applications:

In the current age of data explosion, parallel processing is very much essential for performing a massive volume of data in a timely manner. Parallelization techniques and algorithms are used to achieve better scalability and performance for processing big data. Map reduce is a very popularly used tool or model used in industry and academics. The two major advantages of map reduce are encapsulation of data storage, distribution, replication details. It is very simple for use by the programmers to code for the map reduce task. Since the map reduce is schema free and index free, it requires parsing of each records at the reading point. Map reduce has received a lot of attentiveness in the fields of data mining, information retrieval, image retrieval etc.

The computation becomes difficult to be handled by traditional data processing which triggers the development of big data apps[9]. Big data provides an infrastructure for maintaining transparency in manufacturing industry, which has been having the ability to unreveal uncertainties that exists in the component performance and availability. Another application of the big data is the field of bioinformatics [10] which requires large scale data analysis.

Advantages of big data:

The big data allows an individual to analyze the threats he/she faces internally by naooing onto the entire data landscape over the company using the rich set of tools that the software supporting the big data provides. This is an important advantage of big data since it allows the user to make the data safe and secure. The speed, capacity and scalability of cloud storage provides a mere advantage for the company and organization. Big data even allows the end users to visualize the data and companies can find new business opportunities. Data analytics is one more notable advantage of the big data where in which the individual is allowed to personalize the content or to look and feel the real time websites.

III. CHALLENGES AND DISCUSSIONS

We live in the period of the big data where we can gather more information from daily life of human being. So far, researchers are unable to unify the features that are more essential to big data, many think that big data is something which we cannot process using existing technology, theory or any methods of such kind. However the world has become helpless since enormous amount of data is being generated by science,business and even society. Big data has posed many challenges to the IT industry.

IV. BIG DATA MANAGEMENT:

The needs of the big data are not being satisfied by the current technologies and the speed of increasing storage capacity is much less compared to the data. Thus a revolution reconstruction of information framework is needed very much. For this we need to design a hierarchical architecture for storage. The heterogeneous data are not efficiently handled by the efficient

Algorithms that exist now and thus we need to even design a very efficient algorithm for the effective handling of the heterogeneous data.

Necessity of security in big data:

The big data is used by many of the business but they may not have assets from perspective of the security. If any security threat occurs to big data, it may come out with even more serious issue. Nowadays, companies use this technology to store data of petabyte range regarding to the company, business and customers. This result in severe criticality for classification of information.to secures the data we either need to encrypt, log or use honeypot techniques. The challenge of detecting threats and malicious intruders, must be solved using big data style analysis.

Analysis and computation of big data: Speed is the main thing when we look up for querying in the big data. However the process may be time consuming only because of the reason that it cannot traverse all related data in the whole database in a short time. While the big data is getting complicated, the indices in the big data are aiming at the simple type of the data. The traditional serial algorithm is inefficient for this big data.

V. PROPOSED APPROACHES FOR SECURITY OF BIG DATA

IN CLOUD COMPUTING ENVIRONMENT:

Here we present few security measures that can be used to improve the cloud computing environment.

1) *Encryption*:

Since the data in any syatem will be present in a cluster, a hacker can easily steal the data from the system. This may become a serious issue for any company or organization to safeguard their data. To avoid this, we may go for encrypting the data. Different encryption mechanisms can be used omn different systems and the keys generated should be stored secretly behind firewalls. By choosing this method the data of the user may be kept securely.

2) *Nodes authentication*:

The node must be authenticated whenever it joins the cluster. If the node turns out to be a malicious cluster then such nodes must not be authenticated.

3) *Honeypot nodes*:

The honeypot nodes appears to be like a regular node but is a trap. It automatically traps the hackers and will not allow any damage to happen to the data.

4) *Access control*:

The differential privacy and access control in the distributed environment will be a good measure of security. To prevent the information from leaking we use a SELinux. The Security Enhanced Linux is a feature that provides the mechanism for supporting access control security policy through the use of linux Security modules in linux kernels.

VI. CONCLUSION

This paper gave a description of a systematic flow of survey of the big data in the environment of cloud computing. We discussed about the applications, advantages and challenges faced by big data when used over a cloud computing environment. We proposed few solutions to safeguard the data in the cloud computing environment. In future, the challenges are need to be overcome and make way for the even more efficient use of the big data by the user on a cloud computing environment. It is very much needed that the computer scholars and IT professionals to cooperate and make a successful and long term use of cloud computing nd explore new ideas for the usage of the big data over cloud environment.

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SORTING HELPS TO PROCESS ICEBERG QUERIES VERY EFFECIENTLY

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ABSTRACT: Iceberg queries are used to get a small result set against large volumes of data stored in databases and warehouses. Most of the data mining queries are basically Iceberg queries. The important issue in the Iceberg query evaluation is the AND operation. The current work is based on the idea of reducing AND operations to boost the effectiveness of the Iceberg query by improving its performance. In this paper, an efficient evaluation algorithm for Iceberg query is proposed to find data item pairs by reducing the bitwise AND operations.

Keywords: Iceberg query, Bitmap indices, Priority queue, Threshold.

I. INTRODUCTION

Data and Information play the vital role in the growth of individuals and organization of high repute. It has been the driving force of the 21st century and will continue its role in the ages to come. So, as an individual or an organization to with stand in its domain, to sustain its quality, and be the winner in the ever competing technological realm, one must make an efficient and effective use of the dynamic data and information. It's the proven fact that there are numerous examples recorded in the world history that "if you are not an efficient and effective user of the information, you will be the looser in your generation in fact you may even lose your recognition on the planet over the time". Whether it is a firm in a remote geographical part of the globe or the one that is sprawled its wings and operations in multiple countries, i.e., a multinational Company or an educational institution, or a Research and Development organization involved in active Research, has to depend inevitably on the information and its constantly changing patterns. The types of data and volume in repositories such as databases and data warehouses are growing extremely large. And also storage patterns and accessing techniques are changing proportionately**.** The data plays even a major role in today's business world to withstand their competitors and grow incessantly. The decision makers of the business organizations such as Managers, Business executives, Analysts and data scientists retrieve small amount of the data from larger volumes of repositories. In order to retrieve data efficiently from data bases and data warehouses the data retrieval procedures play a crucial role. The effectiveness of data retrieval techniques depend up on the data specific queries. The data retrieval techniques in use, basically perform the data evaluation, query processing and optimization operations. The query based data retrieval techniques may be heuristic or cost based in nature. For example, the most of the data mining queries are Iceberg (IB) queries which use the hybrid approach. In this work, focuses on the Iceberg queries, a widely used bulk data retrieval method, and improving the query processing time and optimization using bitmap indices.

II. RELATED WORK

Recently, Iceberg query evaluation is attracted more number of researchers, scientists, decision makers. The reason behind this is demand of scalability and efficiency. Always researchers try to find the best methods of computation which takes limited computing resources for large databases [1].The IB Query analysis showing that the IB queries are consuming more time than association rule formation from the datasets [2].So researchers are working continuously to solve the problem of the IB query evaluation in the domain of data warehousing, data mining, and information retrieval systems. As

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their contribution, many novel ideas and techniques have been generated in IB query evaluation process.

Recent years, the compressed BMP index technique is proposed for evaluation of IB queries [8]. The BMP index is built based on bitmap vector which consists of attribute values. The approach is gaining more popularity for column and row oriented databases [3]. In this approach, they are exposed the property of BMP index and invented efficient evaluation strategy of eliminating scanning and processing of bigger database tables [10]. This approach definitely speeds up IB query evaluation. They finished that compressed Bitmap indexing technique [6, 7] is more capable than existing algorithms of IB Query evaluation.

The IB queries are special type of aggregate queries which computes aggregate functions over columns based on user defined threshold value T. These IB queries results gives very useful business information to decision makers. These queries can give an opportunity to minimize its processing and execution time [4, 5].

III. ICEBERG QUERY EVALUATION

The bitmap indices are used in many applications such as database, data warehouse, web mining and information retrieval system applications. A bitmap index is an organization of vertical columns through a bitmap vector [11].

A bitmap index is an array of bits to represent the attributes values of the database. The bitmap index is represented in the form matrix of $m \times n$ *dimensions*, where m is a column that represents attributes values presence in the database and n indicates set of records present for every attribute in the table. The values which are represented in the form column are named as BMP vectors [9]. The collection bits present in every BMP vector holds n bits length, which, if tuple is present in i^{th} row of the bitmap vector, then i^{th} position is 1 and otherwise all the tuple values of the attributes are 0.

In proposed approach, the figure 3.1 represents the unique values of the attributes present in the database. If the attribute value is present in the database table then it is marked as 1 otherwise it is represented as 0. For each individual attribute present in the database has its own BMP vector corresponding to every value of attribute. Following illustrations gives brief description about bitmap indices.

Table 3.1 Attributes and distinct values

	Position Vector								
	0	1	2	3	4	5	...	...	n
P_1	p_1	p_2	p_1	p_3	p_1	p_2	...	...	p_n
P_2	p_1	p_2	p_3	p_2	p_1	p_5	...	...	p_n
...									
P_n	...	...	...	...	...	...	...	...	...

The BMPector is constructed for different values of the attribute based on attribute values position present in database table. Consider the BMP vector generated for the attribute P_1.

P_1: p_1: [101010...... 0],

p_2: [010001......], p_3: [000100...... 0]... p_n: [000000..... 1]

Similarly, BMP vectors for all other attributes are calculated which are shown as a data table in the Figure 3.1a & 3.1b. Continuing with vector is explained in detail by IB query presented in the BMP table.

Table 3.1a: Data Table

	Position Vector							
	0	1	2	3	4	5	6	7
P	p_1	p_2	p_1	p_3	p_1	p_2	p_3	p_2
Q	q_1	q_2	q_3	q_3	q_1	q_3	q_2	q_3

For example P, Q is the attributes of database table and the corresponding tuple value positions are as given below:

Table 3.1b: BMP Indices

P: Attribute	Q: Attribute
P:[p_1, p_2, p_3]	Q:[q_1, q_2, q_3]
P_1:[01000110]	q_1:[01000001]
P_2:[10011000]	q_2:[00110100]
P_3:[00100001]	q_3:[10001010]

The BMP vector is generated using the compressed equality encoding method. Due to occupation of large memory space and consumption of more time for execution of IB query the uncompressed BMP vector generation technique is avoided.

IV.I IB Query Processing by Ordering the BMP Vectors in Priority Queue.

Algorithm 1:

Input: IBqueryprocessingwithorderedPQ (PriorityQueue PQx, PriorityQueue PQy, threshold T)

Output: IB_results.

The count function calculates number of 1 bits present in the bitmap vectors. The PQx and PQy are two priority queues use to store bitmap vectors according priority order.

```
1. PQx = null, PQy = null
2. For each vector x of attribute Y do
3.        xcount = BIT1_COUNT(x)
4.        if xcount >= T then
5. For each vector x of attribute X do
6.        Bubblesort.PQx (x, attribute X )
7. For each vector y of attribute Y do
8. ycount = BIT1_COUNT (y)
9.       if ycount  >=  T then
10. Repeat each vector y of attribute Y do
11.       Bubblesort.PQy (y, attribute Y)
12. R = null
13. While PQx ≠ null and PQy ≠null
14.       x,y= selectvectorpair (PQx, PQy)
15.       Result = BITWISE_AND(x, y)
16. if  Result_count >= T then
17. Add IB_result (x_value, y_value, Result_count) into R
18.       xcount = xcount - Result_count
19.       ycount = ycount - Result_cout
20. if xcount  <  T then
21.       Remove x from PQx
22. if ycount < T then
23.       Remove y from PQy
24. Return R
```

In the first phase of proposed priority queue algorithm, all the values of aggregate attributes X and Y are entered into *priority queue PQx* and *PQy based on initial* high 1's count values (Lines 1 to 11) is employed by in bubble sort algorithm.

The second phase of the algorithm (i.e., Lines 12 to 18) are implemented by selectvectorpair function to process the IB query efficiently by performing bitwise-AND operation between the selected pair of BMP vectors from each Priority Queues (PQ's). After bitwise-AND operation one's count in resultant vector is examined. If the resultant vector value is above T then the vector pair is added to IB result set i.e., *R* and pruning is applied dynamically by decreasing the r's count from original vector. The modified count is compared with threshold, if it is above *T.* It remains in the *PQ* in same position otherwise vector is pruned (Lines 18 through 23). The similar process is conducted continuously until the entire priority queue PQ becomes empty.

The bubble sort algorithm is used in the priority queue algorithm in order to orange the bitmap vectors in sorted order. This algorithm sorts the bitmap vectors according to their 1's count value and these vectors are stored in Priority queue.

Algorithm 2:

```
Algorithm: Bubblesort (long n, long A [])
{
    long a, b, Temp ;


for ( a= 0 ; a < ( n - 1 ); a++)
{
for (b = b+1 ; b< n; b++)
{
if (A[a] > A[b])
{
/* Swapping */
```

```
Temp = A[a];
A[a] = A[b];
A[b] = Temp;
}
}
}
for (a=0, a<=n,a++)
{
Printf ("%d", A[a]);
}
```

The above program indicates the implementation of Bubble sort which is used to sorts the BMP vectors and inserts the BMP vectors with initial high 1's count value into priority queues. The both queues are ordered after completion of the process.

IV.II VALIDATION OF ALGORITHM ICEBERG QUERY PRIORITY QUEUE WITH HIGH 1'S COUNT

The Validation of the IB query is validated on the sample database. The IB query evaluation carried by a SQL query is represented as follows:

Select X, Y, count (*) from R group by X, Y having count (*) > 2;

The above query is used to retrieve the X and Y attribute count values using COUNT function from the database relation 'R' for the user defined threshold T, which is greater than the value 2 should be selected. The IB query that we need to answer is mentioned above, the database table R and BMP indices are those in Table 1a and 1b. The Sample database table R and BMP indices of an aggregate attributes X and Y are represented in the table

As per algorithm the BMP vectors are ordered in priority queues say *PQx* and *PQy* with an initial high 1 bit count values as X_2 *and* X_1 from the BMP index X and Y_1, Y_2 *and* Y_3 from the BMP index Y are selected and stored in two Priority Queues.

In the above representation X_3 *vector* is pruned directly whose 1's count contribution is not satisfying IB threshold T. The two vectors X_2 and Y_1 are selected from *PQx* and *PQy*. After this selection the AND operation is performed between X_2 and Y_1. The resultant vector r is computed by counting the number 1's present in two bitmap vectors as 3 and this count is compared with threshold value which is greater than 2. As the count is passing threshold value T the vector pair X_2 and Y_1 are declared as IB result and the obtained resultant values are added to IB result set. At this movement, 1's count value of two original vectors X_2 and Y_1 are decreased by *r's* count. The X_2 vector 1's count is modified as 4 and Y_1 as 2. Then the 1's count of updated vectors X_2 and Y_1 are compared with threshold *T*.

4.1. Database Table R Bitmap Index 1A Bitmap Index 1B

X	Y	Z	X_1	X_2	X_3	Y_1	Y_2	Y_3
X_2	Y_2	1.23	0	1	0	0	1	0
X_1	Y_3	2.34	1	0	0	0	0	1
X_2	Y_1	5.56	0	1	0	1	0	0
X_2	Y_2	8.36	0	1	0	0	1	0
X_1	Y_3	3.27	1	0	0	0	0	1
X_2	Y_1	9.45	0	1	0	1	0	0
X_2	Y_2	6.23	0	1	0	0	1	0
X_2	Y_1	1.98	0	1	0	1	0	0
X_1	Y_3	8.23	1	0	0	0	0	1
X_2	Y_2	.11	0	1	0	0	1	0
X_2	Y_1	3.44	0	0	1	1	0	0
X_3	Y_1	2.08	0	0	1	1	0	0

With the effect of dynamic pruning the vector Y_1 is pruned and removed from *PQy*, whereas a vector X_2 remains in *PQx* in the same position (because our strategy is not going to rearrange the *PQ* with latest count). Now the vector X_2 is picked up from *PQx* and Y_2 from *PQy*. Then, the AND operation is performed between these two vectors X_2 and Y_2. The resultant vector is examined for 1's count and is computed as 4. Since they are satisfied the threshold value T the vector pair X_2 and Y_2 are declared as IB result and then sent these results to IB result set. In similar to above the vectors X_2 and Y_2 is updated 1's count as 0 and 0 respectively. Hence the vectors X_2 and Y_2 are pruned and removed from *PQx* and *PQy*.

Later the vectors X_1 and Y_3 fetched from *PQx* and *PQy* to conduct bitwise-AND operation. The vectors X_1 and Y_3 are sent to IB result set as the result vector contains more than threshold T. Both priority queues become empty after applying the dynamic pruning on X_1 and Y_3. The algorithm performs a total of three AND operations and produced the result tuples as X_2, Y_1 with count as 3, X_2, Y_2 with count as 4 and X_1, Y_3 as 3.

V.CONCLUSION

In recent years, the bitmap indices are playing very important role in minimizing number of bits present in the bitmap indices. Then, it helps to reduce number of AND operations performed on the bitmap vectors. This reduces greatly the processing time of the query as well as disk space usage. Bitmap based algorithms open several new directions of research. One nice aspect is that they are able to deal efficiently with full logic rules, supporting OR and NOT logical operators in addition to the classical AND. OR is very useful for discovering association rules according to a generalization hierarchy.

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INTRUSION DETECTION SYSTEM USING ARTIFICIAL IMMUNE SYSTEM AND SOFT COMPUTING

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Abstract: **The main focus from one of the problems in computer networks is computer security systems because of the high threat of attack from the internet in recent years. Therefore, an Intrusion Detection System (IDS) that monitors the traffic of computer networks and oversight of suspicious activities in a computer network is required. Several researches have used artificial neural networks combined with a fuzzy clustering method to detect attacks. However, there is an issue that arise from the use of such algorithms. A single artificial neural network can produce overfitting on intrusion detection system output. A number of works in the field of intrusion detection have been based on Artificial Immune System and Soft Computing. Artificial Immune System based approaches attempt to leverage the adaptability, error tolerance, self- monitoring and distributed nature of Human Immune Systems. Whereas Soft Computing based approaches are instrumental in developing fuzzy rule based systems for detecting intrusions.**

Index Terms—Intrusion Detection; Artificial Immune System; Soft Computing.

I. INTRODUCTION

When any set of actions make an effort to compromise with the security properties such as confidentiality, integrity, availability of resources and repudiation then these actions are called intrusions and detection of such intrusions are known as intrusion detection system. The primary objective of IDS is to categories the normal and suspicious activities in the network [26]. The basic functionality of IDS depends on three main components such as data collection, detection and response. The data collection component is responsible for collecting the data from various sources such as system audit data, network traffic data, etc. Detection module is responsible for analysing the collected data to detect the intrusions, and if any suspicious activity detected than initiates the response by the response module.

There are three detection methods presented in the literature such as misuse based, anomaly based and specification based techniques [2, 3, 18]. The first method, misuse based detection systems detect the intrusions on the behalf of predefined attack signature. Second, anomaly- based detection technique detects the intrusion on bases of normal behavior of the system. Defining the normal behavior of the system is a very challenging task because behavior of system can be changed time to time. This technique can detect the unknown or new attacks but with high false positive rates. The third technique is specification - based intrusion detection. This technique specified or defined the set of constraints on a specific protocol and then detects the intrusions at the run time violation of these specifications. Therefore, defining the specification is very time consuming job in this technique.

Normally there are three basic types of IDS architecture in literature: Stand-alone or local intrusion detection systems, Distributed and Cooperative intrusion detection. The proposed system would try to learn the unknown vulnerabilities and make use of those to prepare an adaptive system that would organize itself automatically when such vulnerabilities actually take place. They are computationally intensive and apply machine learning (both supervised and un-supervised) techniques to detect intrusions in a given system. A combination of these two approaches could provide significant advantages for intrusion detection.This paper proposed a novel intrusion detection system based on soft computing techniques for mobile ad hoc networks. The proposed system is based on neuro-fuzzy classifier in binary form to detect, one of vey possible attack, i.e. packet dropping attack.

II. LITERATURE REVIEW

A. Intrusion Detection System

The intrusion detection system is a security protection problem when appear sophisticated and polymorphous intrusion attacks. In general, there are two main types of IDS such as network intrusion detection system (NIDS) and host-based intrusion detection system (HIDS). The anomaly detection approach usually uses statistical analysis and pattern recognition to solve. It is able to detect anomaly intrusion without prior knowledge. Therefore the model has the generalization capability to extract intrusion rule during training.

The statistical anomaly-based IDS determines what bandwidth is generally used, what ports and devices generally connect to each other and alert the manager when the connection is detected which is anomaly or normal [12]. Most approach to build IDS such as Abadeh et al. [13] refer to uses the genetic fuzzy systems (GFSs) hybrid model to solve intrusion attacks problem. They presented three kinds of genetic fuzzy systems based on Michigan, Pittsburgh and iterative rule learning approach to deal with intrusion detection. Alteaijry and Algarny [14] presented a Bayesian based intrusion detection system which based on Bayesian probability theory to filter the intrusion attacks. Horng et al. [15] presented a hierarchical clustering and support vector machines hybrid model to build an IDS. These three researches presented three approaches to solve the same KDD Cup 1999 dataset. However the intrusion detection dataset of KDD Cup 1999 is very popular. In this paper we use the dataset of KDD Cup 1999 to develop our IDS.

B. Artificial Immune Algorithms

In computer science, artificial immune systems are based on of computationally intelligent systems inspired which the principles and processes are simulating to the vertebrate immune system. The AIS are adaptive system, inspired by theoretical immunology and observed immune functions, principles and models, which are applied to problem solving [16]. AIS approach has high performance compared to artificial neural networks, GAs, fuzzy systems and so on, also it has been successfully applied to many fields such as clustering, classification, and pattern recognition, computer defence, optimization, and so on [16]–[22].

C. Soft Computing and Machine Learning

The lack of exactness and inconsistency in the network traffic patterns has encouraged a number of attempts towards intrusion detection based on 'Soft Computing' [11] [12]. 'Soft Computing' techniques attempt to devise inexact and approximate solutions to the computationally-hard task of detecting abnormal patterns corresponding to an intrusion. [4] proposes a Soft Computing based approach towards intrusion detection using a fuzzy rule based system. [15] suggests an approach based on machine learning techniques for intrusion detection applies a combination of protocol analysis and pattern matching approach for intrusion detection.[18] proposes an approach towards intrusion detection by analyzing the system activity for similarity with the normal flow of system activities using classification trees. [10] presents a proactive detection and prevention technique for intrusions in a Mobile Ad hoc Networks (MANET).

III. THE DESIGN

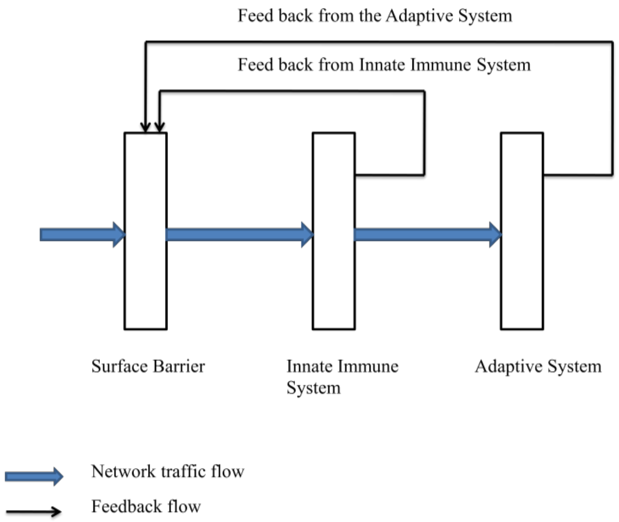

Fig.1. Three primary components of the system

The design of the proposed system is inspired by the Human Immune System; as a result the components of the system have one-one correspondence with the components of the Human Immune System. The system is divided into three primary components:

1) Surface Barrier

2) Innate Immune system

3) Adaptive system

The network traffic entering the system has to pass through each of the three components. Figure.1 shows orientation of the three components and the flow of network traffic through the three components.

For each component we define a probability value *P* which is the probability with which the component detects an intrusion in the system. The value for *P* is determined and improved based on the training of the system as well as the feedback from the previous executions of the system. The system thus exhibits a self-improving adaptive nature. The proposed system goes through the following three phases:

1) Training Phase: during this phase each of the three components of the system are trained using training data set. Results of the training phase enable each of the three components to detect abnormalities and hence intrusion in live production network traffic data.

2) Detection Phase: this phase involves the actual detection of intrusion in the system. The system encounters live network traffic in this phase.

3) Feedback Phase: data encountered during the detection phase is fed back to the system to improve the performance and efficiency of the components of the system. Even though the generation of the feedback to improve the efficiency of the system is mentioned as a separate phase, the generation and assimilation of the feedback data into the system is a continuous process.

IV. PROPOSED METHODOLOGY

Figure 2 shows the proposed intrusion detection system, which consists of two main engines. The clustering engine performs network traffic clustering into the self or non-self-clusters through unsupervised learning techniques. The AIS engine consists of agents that cooperate for intrusion detection. The term "agent" originally comes from Artificial Intelligence (AI) and refers to anything that can view its environment through sensors and act upon that environment using actuators. In this paper, the term agent refers to software agents. Compared to Dal et al.'s [18] work, in our previous work we proposed a distributed model in which we experimented increased performance and efficiency of these IDs as a result of a greater self-improvement rate compare to a centralized structure. This is due to generation of new memory cells and their dynamic synchronization and distribution to all hosts, and thus an enhanced secondary immune response.

The AIS engine trains the primary detectors generated by the negative selection algorithm based on received information from the clustering engine. Moreover, it improves the performance of primary detectors according to the intrusion report analysis from all hosts. In the architecture, the packet pre-processing module is responsible for extracting several attributes from the network traffic to create network flows. These attributes are selected based on the protocol types shown in Table 1

Packets Protocol	***Features***
IP	Source IP Address, Destination IP Address, Time of the First Packet, Time of the Last Packet, Duration
TCP	Source Port Number, Destination Port Number, Number of Packets, Number of SYN Packet, Number of SYN- ACK packet, Number of RST Packet, Number of RST-ACK Packet, Number of FIN-ACK Packet
UDP	Source Port Number, Destination Port Number, Number of Packets
ICMP	#Eco Request, #Eco Reply

Table 1. Network Flow specification for each type of packet

Data Collection and Pre-processing

In the step we want to collect the connection records from internet user to form the training data for learning the IDS. Each connection record is a user connect to service system for provide services according to user requirement. According to classification problem each connection will tag the answer label for supervise learning.

B. Artificial Immune System (AIS) Evolution Flow

We propose AIS IDS for network environment in Fig. 1. There have 6 main steps to learning anomaly intrusion knowledge which are including initial antibody pools, calculate affinity between antibodies and antigen, create new antibody, Clonal Selection mechanism, antibody level mechanism and Negative Selection mechanism

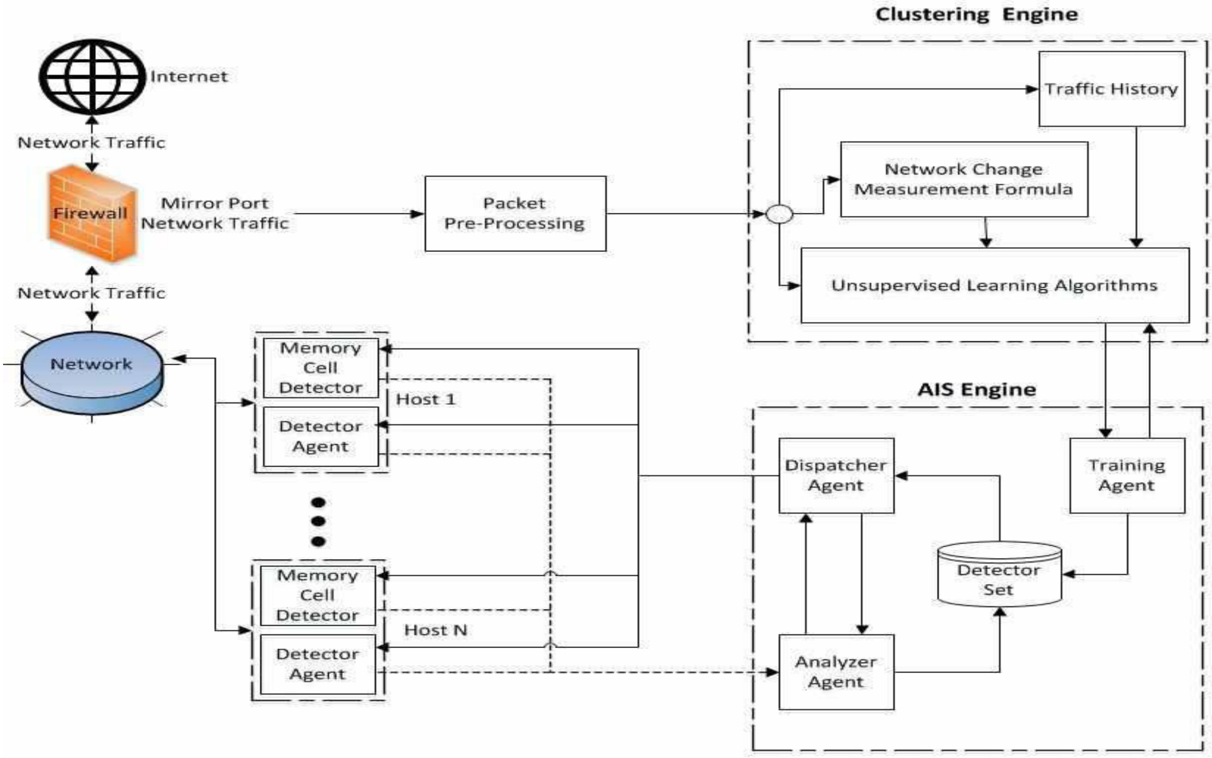

Fig.2. The proposed model

C. Clustering Engine

In order to detect unseen intrusions without using any prior knowledge (training by labeled traffic or signature), we propose a clustering engine as innate immune response. The clustering engine utilizes the DBSCAN clustering method to group the real network traffic into clusters and consider them as self, while behaviors outside of the clusters will be considered as noise or non-self. For this purpose, the engine continuously compares the number of network flows, in different network resolutions (subnets of /0, /8, /16, /24), with a threshold which is dynamically calculated.. Since high speed networks have larger amount of traffic, there is a significant possibility of losing the sign of network attacks. To overcome this issue the system will also monitor the behavior of the network in small resolutions to decrease the possibility of fading the attacks in the normal traffic.

To obtain an accurate threshold, the system needs to determine the previous behavior of the network. It is possible that small attacks to be fade with occurrence of heavy attacks, thus we have applied standardization on the number of network flows by using logarithm (Log) to increase the probability of detecting small attacks during the occurrence of heavy attacks. To determine changes in the network traffic, the system will calculate the “standard deviation” of the number of network flows in different windows from last minute of the traffic. As shown in Table 2 the previous 60 seconds of traffic is divided into four 15 seconds windows.

For instance, □□ is the standard deviation of number of network flows in the first window which is from the last 65 seconds to the last 50 seconds of the previous network traffic. As it has been seen in so many datasets, it takes 2 to 3 seconds from starting time of the network attacks (such as DOS/DDOS attacks) till its own peak.

D. Detect the intrusion for testing

In this part, we want to detect the intrusion attacks which true intrusion attacks then put forth the alert for the network manager just in time to solve anomaly. In this paper we only use the rules of Memory Cell antibody to find which connections is the intrusion. The Memory Cell has high-distinguish capability for most intrusion connections because the three mechanisms can capture perfect antibodies which they high usage rate and high affinity.

V. EXPERIMENTAL RESULTS

The KDD-Cup99 data set from UCI repository [27] is widely used as the benchmark data for IDS evaluation. In our experiments, we random select 49252 records from its 10% training data consisting of 494021 connection records for training. There have 31124 records for testing as in Fig. 3. Each connection record represents a sequence of packet transmission starting and ending at a time period, and can be classified as normal class and 4 different classes of attacks.

(1) Denial-of-service (DOS): Denial of the service that are accessed by legitimate users, e.g., SYN flooding.

(2) Remote-to-local (R2L): Unauthorized access from a remote machine, e.g., password guessing.

(3) User-to-root (U2R): Unauthorized access to gain local super-user (root) privileges, e.g., buffer overflow attack.

(4) Probing (Probe): Surveillance and probing for information gathering, e.g., port scanning.

In model evaluation approach which we use accuracy to measure the classification effectiveness from different classification techniques, as in Equation.

$$\text{Accuracy} = \frac{TP}{TP + TN + FN + FP}$$

where TP denotes true positive;

FP denotes false positive;

FN denotes false negative;

TN denotes true negative.

From the experimental result in TABLE 2, our AIS

IDS has 91.92% accuracy in small case. Our IDS has good performance compare to other machine learning approach.

The proposed methodology was implemented in Java with 40 host Pentium V machines connected to a Server. When the malicious attacks by virus or worms or malware were induced the proposed methodology exhibits the following results as in Fig. 3.

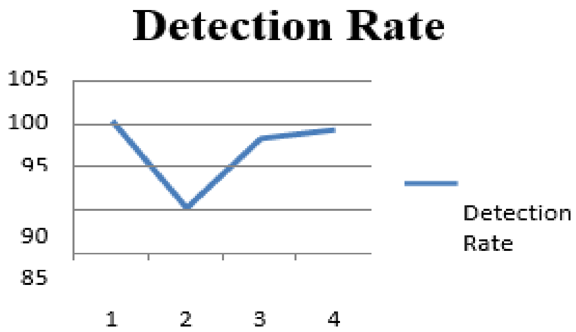

Fig.3 Depicts Detection Rate (%) w. r. t Number of Malicious Input

The malicious files are induced ranging from 1-1000 as inputs on the X-axis whereas the detection rate in % is on the Y-axis. The malicious files are induced in the simulated environment for detection of vulnerability of the proposed methodology.

Number of Input files at a time (X-Axis)	Detection Rate (Y-Axis) %
1	100
10	90
100	98
1000	99

Table 2 Tabular details of the Number of Inputs vs. Detection Rate

VI. CONCLUSION

In conclusion, the suggested hybrid approach based on Artificial Immune System and Soft Computing is instrumental in detecting intrusions and malicious activities in a given network. The three primary components of the system: surface barrier, innate immune system and adaptive system provide a layered defense mechanism, which evolves over multiple executions to combat new emerging attacks. Pattern matching performed by the Innate Immune System using Dynamic Time Warping provides efficient recognition of the self and non-self-patterns in the network traffic data stream. The use of computationally intensive soft computing and machine learning techniques by the Adaptive System, provides additional advantage as far as analyzing complex network traffic data is concerned.

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VIII. ACKNOWLEDGMENT

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IX. ABOUT THE AUTHOR

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QUICK READER TO ENCODE THE QR CODE USING IOS

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Abstract: QR (Quick Response) Code is a two-dimensional barcode that are used to encode and decode information. This encoded data can be decoded by scanning the barcode with a mobile device that is equipped with a camera. QR codes can be used to link to any URL. They can also be used to automatically add information into a user's. Smartphone such as a calendar event, map or personal contact information A QR code is capable of storing several hundred times more information than a conventional barcode and is readable from any direction In this paper, the prior knowledge about A QR Code system is used in combination with a QR Code printer (or QR Code creation software) and QR Code scanner. QR Code is generated with QR Code creation software and a special printer. The QR code Scanner will scan the respected code and provide all the respected information of any project. The details will be such as industry, project team, support team and high level description about the respect client.

Keywords: IOS, QR Code, Generation, recognition.

1. INTRODUCTION

Two-dimensional code which is also called two-dimensional bar code utilizes specific geometrical figures, according to the encoding rules point, empty and white graphics in small area to mark data symbol information. It can express a large amount of information in a very small area. [1] 2D barcode also can express information in the horizontal and vertical so that its storage density is very high. In addition, it has lots of advantages such as good correction capability, expressing information of kinds of figures and characters capability, high privacy, anti-fake and so on[4] .With the increasing of IOS mobile phones' occupancy in mobile internet, the applications of 2D barcode in IOS is wider and wider. Phones can not only be the input terminals of forming 2-dimensional bar code, but also the cameras to sweep and recognize 2-dimensional bar code. Mobile phone 2D barcode has been applied and popularized by major mobile phone manufactures recently. The main content of this paper is about two-dimensional bar code Recognition System which aims at encoding QR Code, whose development bases on IOS

2. RELATED WORK

- Scanning Module
- Detail description Module (About the scanned bar code)
- Team Details Module

Scanning Module

It's important to underline at this point that any barcode scanning, including QR codes, is totally based on video capturing, that's why AV Foundation framework supports the barcode reading feature. Initially, when the app is launched, the interface shown in the figure below is first displayed. By tapping the start scan button, a video capturing session is initiated in order to scan for QR codes. Note that this application cannot be tested on the Simulator, neither to a device without a camera, as everything is based on real-time video capturing. Therefore, you'll need to connect your device and run it there if you want to see the application live. Scan QR codes, and barcodes of all varieties with AV Capture Meta Data Output, new to IOS 7. All you need to do is set it up as the output of an AV Capture Session, and implement the capture Output: did Output Metadata Objects: from Connection.

Fig:-1 QR Scanner

Module

In this module our Scanner will show the Detail description of the Scanned code

Team Details Module

This module results will be the team working on the project all the information about that team

Algorithm

QR code is detected by a 2-dimensional digital image sensor and then digitally analyzed by a programmed processor.

The processor locates the three distinctive squares at the corners of the QR code image, using a smaller square (or multiple squares) near the fourth corner to normalize the image for size, orientation, and angle of viewing. The small dots throughout the QR code are then converted to binary numbers and validated with an error-correcting algorithm.

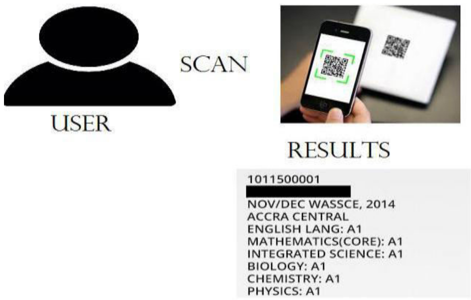

Fig:-2 Project architecture

3. Implementation

- The algorithm with four error correction levels. The higher the error correction level, the less storage capacity.
- Level L (Low) 7% of code words can be restored.
- Level M (Medium) 15% of code words can be restored.
- Level Q (Quartile) 25% of code words can be restored.
- Level H (High) 30% of code words can be restored.
- Due to error correction, it is possible to create artistic QR codes that still scan correctly.
- The format information records two things: the error correction level and the mask pattern used for the symbol. Masking is used to break up patterns in the data area that might confuse a scanner, such as large blank areas or misleading features that look like the locator marks.
- The mask patterns are defined on a grid that is repeated as necessary to cover the whole symbol.

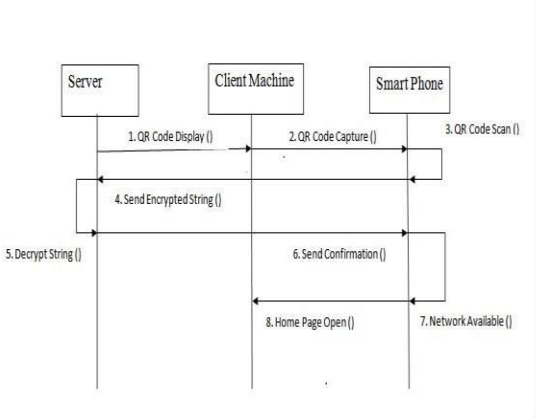

Fig:-3 Application Flow

The important feature, the QR code reading.

A start Stop Reading which is already declared and connected to the “start scan” button item. However, once the video capturing begins, the start button becomes a stop button and vice versa. Programmatically, that means that we need to find a way that lets the app know

when to perform the start functionality, and when to perform the stop functionality. if the app is currently scanning for a QR code and the is Reading flag is YES, then we call the stop Reading method.

Generate & Print QR Code barcode mages in iPhone (iOS) Application

I Phone QR Code Generator SDK is a QR Code generator component designed for iOS project developers who need add QR Code creation features into their developmental apps for iPhone. QR Code generation support on desktops as well as servers. Generate QR Code on iPhone client apps, without communicating with a server complete iPhone QR Code Generator Guide provided with iPhone QR Code generation demo project .QR Code is a matrix code (or two-dimensional bar code) created by Japanese corporation Denso-Wave in 1994. The "QR" is derived from "Quick Response", as the creator intended the code to allow its contents to be decoded at high speed.

QR Code Valid Data Scope IPhone QR Code supports, Numeric data (digits 0 - 9), Alphanumeric data (digits 0 - 9; upper case letters A -Z; nine other characters: space, $ % * + - . / :),Byte data (default: ISO/IEC 8859-1); Kanji characters.

4. EXPERIMENTAL RESULTS

Fig:-4 App Quals Layout

Fig:-5 Detail description layout

Fig:-6 Team Details Module

5. CONCLUSION

QR codes can be used on various mobile device operating systems. QR codes have become common in consumer advertising. Typically, a smart phone is used as a QR code scanner, displaying the code and converting it to some useful form. QR code is its versatility. QR codes can be used for anything and everything. They are also beneficial for both customers and businesses. With the arrival of the big data time, applications of two-dimensional bar code are wider and wider. The QR Code with its accommodating a large amount of information, quick response and efficient representation of Chinese characters, become a mainstream code in matrix two-dimensional code. QR Code will have a wider and wider application prospect in the market of Chinese country. This paper mainly using the open source ZXing library to achieve the generation and recognition of QR Code, Experiments shows that it will have certain promotion effect for Two-dimensional bar code.

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A DECENTRALIZED FRAMEWORK FOR MULTI-OWNER-MULTI-USER SENSOR NETS

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ABSTRACT

Wireless sensor network (WSN) is deployed there's frequently a need to update buggy old small programs or parameters stored within the sensor nodes. This really is frequently accomplished with the data discovery and distribution protocol, which facilities and origin to inject small programs, instructions, queries and configuration parameters to sensor nodes. An info discovery and distribution protocol for wireless sensor systems (WSNs) makes up about upgrading configuration parameters of, and disbursing management instructions to, the sensor nodes. All existing data discovery and distribution techniques are stricken by two drawbacks. First, they are a consequence of the centralized approach only the base station can distribute data item. This type of approach is not suitable for emergent multiword-multi-user WSNs. Second, people techniques were not created using reassurance in your mind and for that reason competitors can easily launch attacks to harm the network. This paper proposes the initial secure and distributed data discovery and distribution protocol named (DiDrip). Many of these disposable sensors might be networked in several programs that require unwatched methods. An Invisible Sensor Network (WSN) includes 100s or thousands of individuals sensor nodes.

INTRODUCTION

Realize that it's completely different from the code distribution techniques which distribute large binaries to reprogram the entire network of sensors. With the sensor nodes might be distributed within the harsh atmosphere, remotely disseminating such small data for that sensor nodes while using wireless funnel could be a more preferred and practical approach than manual intervention. Also we uncover the security vulnerabilities in existing data discovery and distribution protocol [1]. Motivate using the above observation, this paper because the following primary contribution 1 involve distributed data discovery and distribution protocol isn't brand-new, but previous work didn't address this need we see the functional reliance upon such protocol, and pointed out there design objective

RELATED WORK

Several approaches happen to be suggested lately for data discovery and distribution in WSNs. An information discovery and distribution protocol, for wireless sensor systems (WSNs) is answerable for upgrading configuration parameters of, and disbursing management instructions to, the sensor nodes. Most existing research is
dependent on location information that isn't always acquired easily, efficiently and precisely. We advise the idea of Contour-cast, an area-free data distribution and discovery method for large-scale wireless sensor systems. Multidimensional WSNs are deployed in complex conditions to sense and collect data relevant to multiple characteristics (multidimensional data). Such systems present unique challenges to data distribution, data storage as well as in-network query processing (information discovery). We present simulation results showing the perfect routing structure is dependent around the frequency of occasions and query occurrence within the network. All existing data discovery and distribution methods undergo from two drawbacks. First, they derive from the centralized approach just the base station can distribute data item. Wireless sensor systems (WSN) are attractive for information discovery in large-scale data wealthy conditions and may increase the value of mission-critical programs for example fight-field surveillance, ecological monitoring and emergency response. However, to be able to fully exploit these systems for such

programs. Among the primary issues in acquiring multicast communication may be the source authentication service. Sensor systems deployed in hostile areas are susceptible to node replication attacks, by which an foe compromises a couple of sensors, extracts the safety keys, and clones them in a lot of replicas, that are introduced in to the network to do insider attacks. Data distribution and discovery is crucial for ad-hoc wireless sensor systems. Additionally, it balances push and pulls procedures in massive systems enabling significant QoS enhancements and savings [2]. Multicast communication has become the foundation for an increasing number of programs. Therefore, acquiring multicast communication is really a proper requirement of effective deployment of huge scale business multi-party programs.

III. METHODOLOGY

All recommended techniques believe that the operating atmosphere inside the WSN is reliable and includes no foe. However, the truth is, competitors exist and impose risks for your normal operation of WSNs. This issue has only been addressed recently by which is recognized the security vulnerabilities of Drip and proposes an effective solutions. More to the stage, all existing data discovery and distribution techniques employ the centralized approach. Sadly, this process is battling with the only real reason for failure as distribution does not appear possible when the base station is not functioning or when the link between the underside station and a node is broken. Additionally, the centralized approach is inefficient, non-scalable, and susceptible to security attacks which may be launched anywhere inside the communication path. A concealed sensor network (WSN) includes spatially distributed autonomous sensors to look at physical or environmental conditions, for instance temperature, appear, pressure, etc. and also to cooperatively pass their data while using the network getting a principal location. DiDrip includes four phases, system initialization, user joining, and packet pre-processing and packet verification. For the fundamental protocol, in system initialization phase, the network owner produces its public and private keys, then loads everyone parameters on every node before the network deployment. In user joining phase, you receive the distribution privilege through signing up towards the network owner. In packet pre-processing phase, just in case your user can get into for your network and requires to distribution some data items, he/she'll have to make the data distribution packets then send people for the nodes. In packet verification phase, a node verifies each received packet. Whether or not this seems sensible positive, it updates the data while using received packet. While using the design objectives, they propose DiDrip [3]. It is the first distributed data discovery and distribution protocol, which allows network entrepreneurs and approved clients to disseminate data items into WSNs without depending over the base station. Furthermore, our extensive analysis indicates that DiDrip satisfies the security needs inside the techniques available. Particularly, they normally use the provable security approach to formally prove the authenticity and integrity inside the disseminated data items in DiDrip. The higher modern systems are bi-directional, also enabling control of sensor activity. The development of wireless sensor systems was motivated by military programs for instance battleground surveillance today such systems be employed in several industrial and consumer programs, for instance industrial process monitoring and control, machine health monitoring, and so on. Due to recent technological advances, the manufacturing of small, affordable sensors elevated to acquire technically and economically achievable. The sensing electronics measure ambient conditions connected while using the atmosphere all over the sensor and transform them into an electric signal. Processing this kind of signal uncovers some characteristics about objects situated and/or occasions happening near the sensor. Several of these disposable sensors might be networked in many programs that require unwatched techniques. A Concealed Sensor Network (WSN) includes 100s or thousands of people sensor nodes [4]. These sensors be capable of communicate either among each other in order to an exterior base-station (BS). More sensors allows for sensing over bigger physical regions with greater precision. Formerly few years, an extensive research that addresses the risk of collaboration among sensors in data gathering and processing along with the coordination and control of the sensing activity were moved out. However, sensor nodes are restricted in energy supply and bandwidth. Thus, innovative techniques that eliminate energy inefficiencies which will shorten the time-frame in the network are highly needed. Such constraints combined through getting an average deployment of countless sensor nodes pose many challenges for your design and control of WSN's and needed energy awareness whatever the layers in network protocol stack. We've seen the routing techniques concerning multiple pathways as opposed to just one path in order to raise the network performance. that fault tolerance within the protocol is measured while using likewise the alternate path might be acquired within the source plus a destination when the primary path fails this is often frequently evaluated keeping multiple paths in regard to the origin combined to the destination the free for the evaluated energy consumption and growing customer count.

CONCLUSIONS

DiDrip includes four phases system initialization ,user joining add packet pre processing and packet verification during this paper we've recognized the safety vulnerabilities within the data discovery and distribution when present in WSN's which were not addressed in formerly research. Several data discovery and distribution techniques are really suggested but undertake and don't of people approaches support distributed operation. Therefore during this paper a great distributed and data discovery and distribution protocol name DiDip is remains suggested.

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ENHANCING THE NETWORK LIFETIME IN WIRELESS SENSOR NETWORKS THROUGH CASER PROTOCOL

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ABSTRACT—*At present wireless sensor networks are suffering from two major problems which are, network lifetime optimization and security. In this paper, we propose a secure and efficient Cost-Aware SEcure Routing (CASER) protocol. By using this protocol we can overcome these conflicts in the wireless sensor networks. The CASER protocol overcomes these two issues through two adjustable parameters: 1. Energy Balance Control (EBC) and Probabilistic random walking. We then find out that the energy consumption is severely asymmetrical to the uniform energy deployment for the specified network topology, which very much reduces the lifetime of the sensor networks. To overcome this problem, we propose an efficient non-uniform energy deployment strategy to enhance the lifetime and message delivery ratio under the same energy resource and security condition. We also provide a quantitative security analysis on the proposed routing protocol.*

1. INTRODUCTION

The modern technological enhancements produce Wireless Sensor Networks (WSNs) technically and economically low cost to be extensively used in every military and civilian applications, like perceptive of shut circumstances related to the setting, costly species and exacting infrastructures. A key feature of such networks is that each network contains sizable quantity of untied and unattended device nodes. These nodes usually have really restricted and non- replenishable energy resources, that produces energy a significant vogue issue for these networks. Routing is that the very important vogue concern for WSN. a straight forward routing protocol provides less energy depletion for message delivery and has the great message delivery magnitude relation. to increase the device network amount of your time and to boot manage total device network energy depletion. Wireless device Networks have the solutions that maintain intensive vary of applications. Supported the appliance, their WSN setting is that the risky, troublesome and fewer problematic. Even the Encoded Security Systems in WSNs to not notice the node physical internment, the malicious or selfish nodes. So, novel security systems unit of measurement essential for the secure routing of message from offer to sink node of WSNs. a singular technique of getting security whereas not victimization cryptography is Trust based totally security in WSNs. Trust is “The mark of Trustworthiness”. It holds the nodes information and collects all record regarding the node. It’s going to check the other node in doing actions and could be formed by keeping a details of the communications with the other nodes directly moreover as indirectly. By victimization these details a trust worth area unit verify. to stay up the selection making processes of the network in wireless device network Trust management area unit used. These help to the removal of the unsecured nodes among the WSN regarding the approaching actions of other nodes (trustees). Several examines on trust associated in WSN unit of measurement done, but it's totally necessary to vogue and develop a trust management system that uses the lesser amount of resources of the node in valuation and management of religion between/among the nodes. The trust management of the WSN would be a straight forward. It not have the constraints on energy consumption, software, hardware, memory usage, computing, method speed and communication metric, and it confirm varied attacks merely, and disease and alter trust relations supported it. Routing can be a attention-grabbing task in WSNs as a results of the restricted resources. Geographic routing was for the foremost half discovered as is that the necessary approach for WSNs. Geographic routing protocols uses the relative position details and send the packets one once another from the availability node to the sink node. Based on the direction or distance the availability node selects future adjacent node and send the message. By victimization signal strength or victimization GPS receivers unit of measurement understand the total distance between adjacent nodes. The details regarding the nearer one in each of next nodes could also be modified between adjacent nodes. Amount of your time plays a key role and to boot major issue in Wireless device networks. In a secure and routing strategy was got the shortest approach. Always selecting and shrewd very cheap energy node and conjointly the approach are reduced by the secured routing strategy among the wireless device networks. AODV or directed diffusion is to boot a reactive protocol to route multiple ways as a result of the secure routing strategy. Then, the routing strategy can opt for a route supported a probabilistic technique per the remaining energy. Among the transmitter power level adjusted per the area between the transmitter and also the receiver. Routing was communicated as a mathematics downside of adjacent node choice to produce best use of the network amount of your time. Then examined the uneven energy consumption for consistently organized data-assembly device networks. The network is split into many regions and each node can produce info aggregation. In routing strategy was planned to stabilize the energy consumption between the nodes at intervals each grid. In formulated the combined approach of finding the routes and conjointly the traffic load allocation, and conjointly the network amount of your time values can increase by the sleep designing. By victimization this could be the concept of opportunist routing, developed a routing metric to report every link trustiness and node remaining energy. The device node calculates the foremost effective metric worth throughout a localized area to appreciate every trustiness and womb-to-tomb maximization.

2. RELATED WORK

Routing may be a difficult task in WSNs attributable to the restricted resources. Geographic routing has been wide viewed as one of the foremost promising approaches for WSNs. Geographic routing protocols utilize the geographic location information to route knowledge packets hop-by-hop from the source to the destination. The supply chooses the immediate neighboring node to forward the message supported either the direction or the gap. The gap between the neighboring nodes is calculable or acquired by signal strengths or mistreatment GPS equipments. The relative location info of neighbor nodes will be changed between neighboring nodes. A Geographic Adaptive Fidelity (GAF) routing scheme was planned for detector networks equipped with low power GPS receivers. In GAF, the network space is divided into mounted size virtual grids. In every grid, only one node is chosen because the active node, whereas the others will sleep for a amount to save lots of energy. The detector forwards the messages supported greedy geographic routing strategy. A question primarily based Geographic and Energy Aware Routing (GEAR) was planned. In GEAR, the sink node disseminates requests with geographic attributes to the target region rather than mistreatment flooding. Every node forwards messages to its neighboring nodes supported calculable cost and learning value. The calculable value considers each the gap to the destination and also the remaining energy of the detector nodes. Whereas the educational cost provides the change info to contend with the local minimum drawback. While geographic routing algorithms have the benefits that each node solely must maintain its neighboring information, and provides a better potency and a more robust scalability for giant scale WSNs, these algorithms might reach their native minimum, which may end in dead finish or loops. To solve the native minimum drawback, some variations of these basic routing algorithms were planned, including GEDIR, MFR and compass routing rule. The delivery magnitude relation is improved if every node is conscious of its two-hop neighbors. There are a couple of papers mentioned combining greedy and face routing to resolve the native minimum drawback. The essential plan is to line the native topology of the network as a flattened graph, and so the relay nodes attempt to forward messages on one or probably a sequence of adjacent faces toward the destination. Lifetime is another space that has been extensively studied in WSNs. A routing theme was planned to find the sub-optimal path that may extend the period of time of the WSNs rather than perpetually choosing all-time low energy path. Within the planned theme, multiple routing methods are ready ahead by a reactive protocol like AODV or directed diffusion. Then, the routing theme can select a path primarily based on a probabilistic technique in keeping with the remaining energy. Yangtze River and Tassiulas assumed that the transmitter power level is adjusted in keeping with the distance between the transmitter and also the receiver. Routing was developed as a applied mathematics drawback of neighboring node choice to maximize the network period of time. Then Zhang and Shen investigated the unbalanced energy consumption for uniformly deployed data collecting sensor networks. During this paper, the network is divided into multiple corona zones and every node will perform data aggregation. A localized zone-based routing scheme was planned to balance energy consumption among nodes at intervals every corona. Liu et al. developed the integrated style of route choice, traffic load allocation, and sleep planning to maximize the network lifetime. Supported the idea of expedient routing, developed a routing metric to deal with each link reliableness and node residual energy. The detector node computes the best metric price during a localized space to achieve each reliableness and lifelong maximization.

3. FRAMEWORK

In our theme, the network is equally divided into little grids. Every grid incorporates a relative location supported the grid info. The node in every grid with the best energy state is chosen because the head node for message forwarding. additionally, every node within the grid can maintain its own attributes, as well as location info, remaining energy state of its grid, further because the attributes of its adjacent neighboring grids. The data maintained by every sensing element node are updated sporadically. We have a tendency to assume that the sensing element nodes in its direct neighboring grids are all inside its direct communication vary. We have a tendency to additionally assume that the entire network is totally connected through multi-hop communications. Whereas increasing message supply location privacy and minimizing traffic jam for communications between the supply and therefore the destination nodes, we are able to optimize the sensing element network period through balanced energy consumption throughout the sensing element network. Additionally, the controlled energy levels of its adjacent neighboring grids are often wont to notice and strain the compromised nodes for active routing choice.

A. CASER Protocol

WSNs routing is usually earth science primarily based, an earth science primarily based secure and economical Cost-Aware SEcure routing (CASER) protocol for WSNs with none result on traffic in network. The protocol agrees messages to be communicated mistreatment the secure routing ways in which, irregular approach and settled routing, based on the correctness. The sharing of those 2 methods is resolute by the definite security necessities. This strategy is analogous to distributing North American country Mail through USPS: send mails value quite regular mails; but, mails are often sending fastly and it provides deliver report. The protocol provides a security and privacy message transmission option to improve the message delivery magnitude relation by the varied and totally different attacks. This protocol has 2 main benefits:

1. It improves the period of total sensors within the grid and decrease the wastage of energy.

2. This protocol helps numerous routing schemes supported the routing necessities. It checks whether or not the message delivering is sweet or not and additionally this protocol will scale back the various attacks like trace back attacks, jamming and etc., in WSNs. Main methods of this paper are often précised as follows:

i) A secure and economical Cost-Aware SEcure Routing (CASER) protocol for WSNs. during this protocol, routing methods are always useful to maintain the message delivery necessities.

ii) A quantitative theme maintains the energy depletion then the sensing element network period and therefore the total range of messages that can be delivered area unit improved beneath a similar energy usage.

3) Developed theoretical formulas to assessment sum of routing hops in protocol supported dynamical routing energy balance management and security necessities.

4) To look at security of the routing algorithmic program.

5) Provided a best non-uniform energy readying strategy for the given sensing element networks supported the energy depletion ratio.

B. Secure and Efficient Routing Strategy (SERS)

The projected Secure and Efficient Routing (SERS) Strategy uses the relative node approach. The SERS is employed relative location for routing. Otherwise to calculate the amount of hops to cut back flooding and varied attacks. This strategy is nice to discover routing attacks and offers the efficient support of wide device networks. Relative location causation is basically protected that opposes on varied attacks to causation the message calculation, have the additional preference for secure routing its node ID and attributes. And conjointly on top of characteristics square measure same for all relative location routing ways like the Greedy Perimeter Stateless Routing (GPSR). SESR includes successive adjacent node choice. It provides the situation of a node, low energy usage and trust or secure message routing. Energy managing is vital for decreasing all the attacks with high trust calculated price. The limiting issue is determined by the nodes energy and routing trust calculated price are going to be reduced i.e. the possibility to finish all the overall work. Due to the calculation, we've incorporated the energy calculations within the total trust price a node computes all of their adjacent nodes. Energy management permits load equalization. It plays terribly crucial to cut back the traffic analysis attacks and increase the network life price. Within the SERS each and every time an occurrence performed supported the trust metrics (packet forwarding, network Acknowledgement) happens, the consistent results placed within the trust table or uninterrupted trust table square measure updated. At identical, whenever name response messages are taken, their info is placed within the indirect trust table or uninterrupted Trust table when the checking. Each table has info for every adjacent node. The whole node's trust price is calculated once a brand new message has got to be sent, even supposing the trust values associate degreed name information is updated each and every time an occurrence possesses or a name response message has been taken. This event-driven theme was adopted to balance the energy and conjointly maintaining the resources. It calculates each interrupted and interrupted trust and it takes as a complete trust price and it considers the space of adjacent nodes. The ultimate price calculated and conforms whether or not the node sends the data or not.

C. SERS for Identifying and Avoid the Malicious Nodes

To evaluate the effectiveness of the secured and efficient routing strategy of the attendance of problematic nodes, first to analyze the strategy with the totally malicious nodes place is implemented in that the relative location node routing is adapted. Connection is jammed when it have malicious nodes on that effective SERS strategy the malicious nodes will be detached and stopped. The Trust of any node is assessed by using the geometric mean of trust metrics of the node (uninterrupted trust), and complete mean of data created on all the adjacent (neighbor) nodes. The node's information given by adjacent nodes (interrupted trust) is the direct trusts with the node. All the nodes in the network, have to keep a database list i.e., what they are doing for every of its adjacent node. This database listen closes the details about the various trust values, i.e. Quality Of Service etc are the features for all its adjacent nodes based on the how many times their attempts performed in the network. This trust values information have to use for computing the uninterrupted trust of its each adjacent node. To calculate the distance of the adjacent node information to the work station, the related calculations joined in the routing function is done as below:

$$T_d^{A,B} = 1 - \frac{d_i}{\sum d_i}$$

Here,

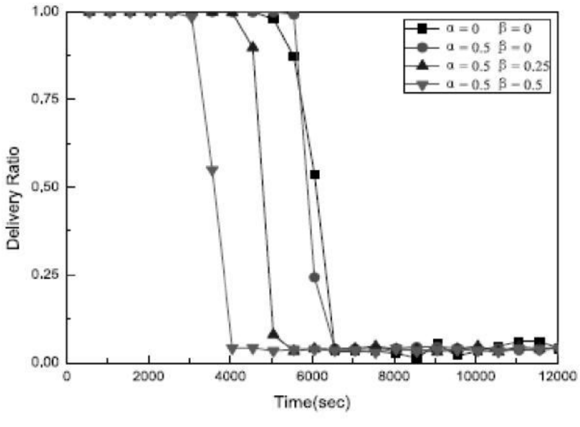
di is the distance between the adjacent node i to the work station and

Σdi represent the distance between the adjacent nodes to the base station.

The above equation gives minimum distance and improves the best value.

4. EXPERIMENTAL RESULTS

In our experiments, we implement balanced energy consumption for all sensor nodes so that all sensor nodes will run out of energy at about the similar time. This design guarantees a high message delivery ratio until energy runs out from all possible sensor nodes at about the same time. Then the delivery ratio drops sharply. This has been confirmed through our simulations.



Above figure shows that delivery ratio under the different Energy Balanced Control (EBC) and Security level.

Finally, our analysis and simulation results represent that we can increase the lifetime and the number of messages that can be delivered under the non-uniform energy deployment by more than four times.

5. CONCLUSION

In this paper, we proposed a secure and efficient CASER protocol for wireless sensor networks. By using this protocol we can balance the energy consumption and reduce network lifetime optimization. CASER has the elasticity to support multiple routing schemes in message forwarding to enhance network lifetime while improving routing security.

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ASSOCIATIONALIGNED WITH MULTIPLEXING: MULTICAST SCHEDULING ALGORITHMS INTENDED FOR OFDM RELAY NETWORKS

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ABSTRACT

Relay-enabled wireless networks (eg. WIMAX 802.16j) stand for correlate increasing trend for the incorporation of multi-hop networking explanations for last-mile broadband access in after that generation wireless networks. The adoption of additional refined access technologies like OFDM (orthogonal frequency division multiplexing) not to mention the relay-induced two-hop nature, provides 2 key edges to those networks within the style of diversity and spatial utilize gains. However, leverage these edges incorporate additional refined solutions, among that, user planning forms a key element. We have a tendency to take into account the particular drawback of planning users with finite buffers on the multiple OFDM carriers (channels) over the 2 hops of the relay-enabled network. We have a tendency to propose planning algorithms that facilitate leverage diversity and spatial utilize gains from these networks. We have a tendency to show that even the planning drawback to take advantage of diversity gains alone is NP-hard and supply each in theory and much economical polynomial-time algorithms with approximation guarantee. Building on the range solutions, we have a tendency to conjointly propose associate economical polynomial-time planning algorithmic program for exploiting each spatial utilize additionally as diversity. The projected solutions area unit evaluated to focus on the relative significance of diversity and spatial utilize gains with relevance varied network conditions.

INTRODUCTION

The last decade has seen a big quantity of analysis in multi-hop wireless networks (MWNs) [1, 2]. Whereas their fully localized nature has contributed to scalable solutions, they need additionally moon-faced important challenges in moving towards business adoption. However, with following generation wireless internet Permission to form digital or exhausting copies of all or a part of this work for private or schoolroom use is granted while not fee only if copies square measurenot created or distributed for profit or business advantage which copies bear this notice and also the full citation on the primary page. To repeat otherwise, to republish, to post on servers or to spread to lists, needs previous specific permission and/or a fee.

Works moving towards smaller (micro, pico) cells for providing higher knowledge rates, there's a revived interest in MWNs from the attitude of integration them with infrastructure wireless networks. With a decrease in cell size, relays square measure currently required to supply extended coverage, leading to a multi-hop network. A two-hop relay-enabled wireless network forms a very important commencement towards such a network model (Figure 1(a)). Here, the booster amplifier (RS) square measure connected to the wireless infrastructure (base station, BS) and supply improved coverage and capability to many applications as well as serving mobile users (MS) in business hotspots, workplace buildings, transportation vehicles, coverage holes, etc. The excess of visualized applications has additionally crystal rectifier to their adoption because the necessary network model in IEEE 802.16j modification to the WIMAX commonplace and successively forms the context for our work. Orthogonal frequency division multiplexing (OFDM) has become the popular selection for air interface technology in future native and wide space wireless networks. The whole spectrum is split into multiple carriers (sub-channels), resulting in many physical layers and planning edges [3, 4]. The 2-hop network model including OFDM provides two key edges, particularly diversity and spatial utilize gains. 3 sorts of diversity gains is exploited through scheduling: (i) multi-user diversity : for a given sub-channel, totally {different|completely different} users expertise different weakening statistics, permitting U.S. to choose a user with a bigger gain; (ii) channel diversity : sub-channels experiencing high gain may vary from one user to a different, allowing multiple users to be allotted their best channels in tandem; and (iii) cooperative diversity : relays will exploit wireless broadcast advantage to collaborate and improve the SNR (signal-noise ratio) at the meant receiver. Additionally to the variety gain, the two-hop network model additionally provides space for spatial utilize, whereby synchronic transmissions on the relay hop (BS-RS) and access hop (RS-MS) is leveraged on constant channel as long as there's no mutual interference. User and channel diversity gains, offered in typical one-hop cellular networks, are effectively leveraged to boost system performance through many channel-dependent planning schemes [3, 4, 5]. However, they are doing not offer spatial utilize or cooperative diversity gains. MWNs on the opposite hand, offer spatial utilize. However, since diversity gains need channel state feedback from RS and MS and should be exploited at fine time scales (order of frames), they can't be effectively leveraged in a very giant multi-hop setting. Relay-enabled networks with a two-hop structure, offer a novel middle-ground between these 2 networks, providing U.S. access to a large number of diversity and spatial utilize gains. Whereas this provides potential for important performance improvement, it additionally requires a lot of refined, tailored planning solutions that take under consideration the two-hop nature of the system. during this context, we tend to concentrate on the particular drawback of planning users with finite buffers on multiple OFDM sub-channels over the 2 hops of the network, whereas with efficiency exploiting the offered diversity and spatial utilize gains. Investing simply the variety gains over 2 hops is a very important drawback in itself, whose hardness we first establish. Then we make the subsequent contributions.

•We provide 2 in theory economical polynomial-time programming algorithms for exploiting diversity gains with approximation guarantees of $\left(1 - \sqrt{\frac{c'K}{N} \cdot \log(2N)}\right)$ and $\left((1 - \frac{1}{e})^2 - \epsilon\right)$ within the order of skyrocketing time-complexity. K and Narea unit the quantity of users and sub-channels severally, and $c' > 1, \epsilon > 0$ area unit constants. We tend toadditionally give a much economical accommodative programming formula with smart average case performance.

• Building on the variety solutions, we tend to additionally propose AN economical polynomial-time programming formula for exploiting each spatial reprocess and variety.

• The planned solutions area unit evaluated to focus on the relative significance of diversity andspatial reprocess gains with relevance varied network conditions. The remainder of the paper is organized as follows. The system description is conferred. The planned programming algorithms for exploiting diversity and special reprocess severally.. Finally, final remarks area unit conferred.

RELATED WORK

Several works [6, 7] have investigated the potential of relay enabled wireless networks to supply improved coverage and capability. These networks have gained attention each within the standards community (IEEE 802.16j) and within the business [8]. Scheduling, being a key element in these networks, has received higher stress [9, 7, 10]. However, most of those works [9, 7] specialize in TDMA variants wherever the planning call reduces primarily to deciding whether or not to use a relay or not and that explicit user. They specialize in link level performance and don't exploit spatial utilize and variety across the relay and access hops that area unit on the market at a network level. Further, they [9, 7, and 10] don't take into account a multiple channel OFDM network (channel diversity), that complicates planning selections with the likelihood of multiple user's operative in parallel. The works on OFDM planning in typical cellular systems [3, 4] can't be directly applied to two-hop relay networks, wherever the network structure is completely different and spatial utilize and variety across hops forms a vital element. There are some works [11, 12] that have checked out multiple channels within the presence of relays, wherever assignment of channels at the second hop is taken into account to use diversity higher. However, the channel assignment model thought-about is simple and additionally spatial utilize isn't exploited. On the opposite hand, works in MWNs [1, 2] leverage spatial utilize, however they are doing not specialize in OFDM and associated diversity gains thanks to the big scale. Also, none of the on top of works take into consideration the finite (non back logged) user buffers at the Bachelor of Science, that makes the matter NP-hard. This was recently thought-about in [5] within the single hop context. We tend to take into account investing each diversity and spatial utilize gains within the presence of finite user buffers during a tougher two-hop setting.

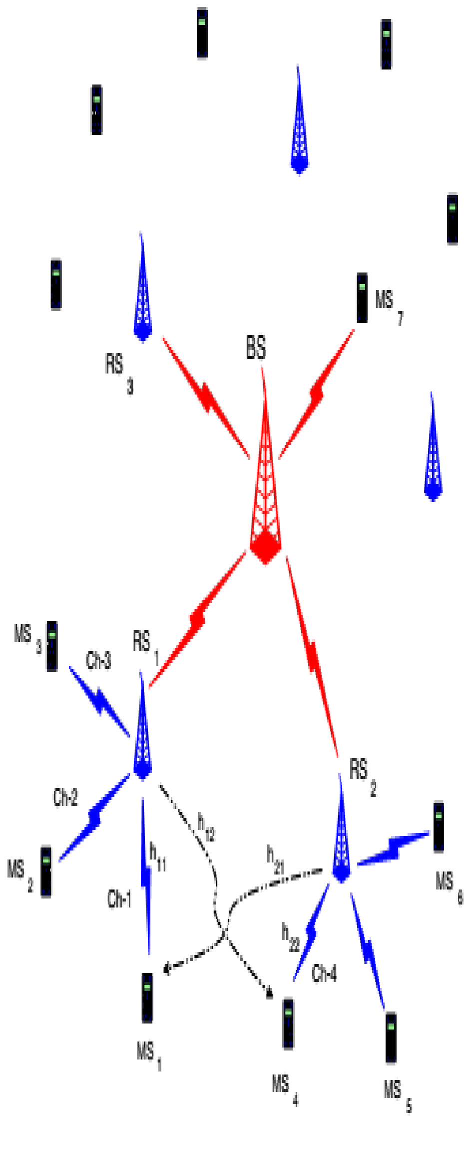
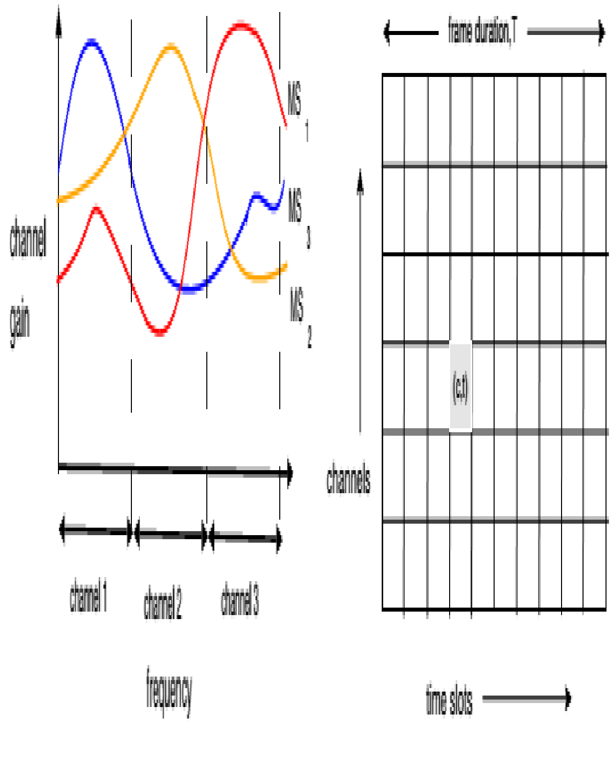


Figure 1: Network Model



a) Multi-user and Channel Diversity b) Frame Structure

Figure 2: Potential Gains and Frame Structure.

CONCLUSIONS

We have thought of the precise drawback of programming user traffic on the multiple OFDM sub-channels over the 2 hops of the relay-enabled wireless networks. We tend to projected programming algorithms that facilitate leverage the variety and spatial use gains from these networks. We tend to showed that even the programming drawback to take advantage of diversity gains alone is NP-hard and provided each on paper and much economical polynomial-time algorithms with approximation guarantees. We tend to additionally projected Associate in nursing economical polynomial time programming algorithmic program for exploiting each spatial use in addition as diversity. Evaluations of the projected solutions highlighted the relative significance of varied diversity and spatial use gains with reference to varied network conditions.

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DECENTRALIZED ACCESS CONTROL WITH ANONYMOUS AUTHENTICATION FOR DATA SECURITY IN CLOUDS

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***Abstract*: We propose a new decentralized access control scheme for secure data storage in clouds that supports anonymous authentication. In the proposed scheme, the cloud verifies the authenticity of the series without knowing the user's identity before storing data. Our scheme also has the added feature of access control in which only valid users are able to decrypt the stored information. The scheme prevents replay attacks and supports creation, modification, and reading data stored in the cloud. We also address user revocation. Moreover, our authentication and access control scheme is decentralized and robust, unlike other access control schemes designed for clouds which are centralized. The communication, computation, and storage overheads are comparable to centralized approaches.**

Index Terms: Access control, authen-tication, attribute-based signatures, attribute-based encryption, cloud storage

INTRODUCTION

Research in cloud computing is receiving a lot of Attention from both academic and industrial worlds.

In cloud computing, users can outsource their computation and storage to servers (also called clouds) using Internet. This frees users from the hassles of maintaining resources on-site. Clouds can provide several types of services like applications (e.g., Google Apps, Microsoft online), infra-structures (e.g., Amazon's EC2, Eucalyptus, Nimbus), and platforms to help developers write applications (e.g., Amazon's S3, Windows Azure).

Much of the data stored in clouds is highly sensitive, for example, medical records and social networks. Security and privacy are, thus, very important issues in cloud computing. In one hand, the user should authenticate itself before initiating any transaction, and on the other hand, it must be ensured that the cloud does not tamper with the data that is outsourced. User privacy is also required so that the cloud or other users do not know the identity of the user. The cloud can hold the user accountable for the data it outsources, and likewise, the cloud is itself accountable for the services it provides. The validity of the user who stores the data is also verified. Apart from the technical solutions to ensure security and privacy, there is also a need for law enforcement.

Cloud servers prone to Byzantine failure, where a storage server can fail in arbitrary ways. The cloud is also prone to data modification and server colluding attacks. In server colluding attack, the adversary can compromise storage servers, so that it can modify data files as long as they are internally consistent. To provide secure data storage, the data needs to be encrypted. However, the data is often modified and this dynamic property needs to be taken into account while designing efficient secure storage techniques.

Efficient search on encrypted data is also an important concern in clouds. The clouds should not know the query but should be able to return the records that satisfy the query. This is achieved by means of searchable encryption. The keywords are sent to the cloud encrypted, and the cloud returns the result without knowing the actual keyword for the search. The problem here is that the data records should have keywords associated with them to enable the search. The correct records are returned only when searched with the exact keywords.

Security and privacy protection in clouds are being explored by many researchers. Wang addressed storage security using Reed-Solomon erasure-correcting codes. Authentication of users using public key crypto-graphic techniques has been studied in. Many homomorphic encryption techniques have been suggested to ensure that the cloud is not able to read the data while performing computations on them. Using homomorphic encryption, the cloud receives ciphertext of the data and performs computations on the ciphertext and returns the encoded value of the result. The user is able to decode the result, but the cloud does not know what data it has operated on. In such circumstances, it must be possible for the user to verify that the cloud returns correct results.

Accountability of clouds is a very challenging task and involves technical issues and law enforcement. Neither clouds nor users should deny any operations performed or requested. It is important to have log of the transactions performed; however, it is an important concern to decide how much information to keep in the log. Accountability has been addressed in Trust Cloud

OUR APPROACH

Efficient search on encrypted data is also an important concern in clouds. The clouds should not know the query but should be able to return the records that satisfy the query. This is achieved by means of searchable encryption The keywords are sent to the cloud encrypted, and the cloud returns the result without knowing the actual keyword for the search. The problem here is that the data records should have keywords associated with them to enable the search. The correct records are returned only when searched with the exact keywords.

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It is just not enough to store the contents securely in the cloud but it might also be necessary to ensure anonymity of the user. For example, a user would like to store some sensitive information but does not want to be recognized. The user might want to post a comment on an article, but does not want his/her identity to be disclosed. The user should be able to prove to the other users that he/ she is a valid user who stored the information without revealing the identity. There are cryptographic protocols like ring signatures mesh signatures group signatures which can be used in these situations. Ring signature is not a feasible option for clouds where there are a large number of users. Group signatures assume the preexistence of a group which might not be possible in clouds. Mesh signatures do not ensure if the message is from a single user or many users colluding together. For these reasons, a new protocol known as attribute-based signature (ABS) has been applied. ABS was proposed by Maji in ABS, users have a claim predicate associated with a message.

RELATED WORK

Access control in clouds is gaining attention because it is important that only authorized users have access to valid service. A huge amount of information is being stored in the cloud, and much of this is sensitive information. Care should be taken to ensure access control of this sensitive information which can often be related to health, important documents (as in Google Docs or Dropbox) or even personal information (as in social networking). There are broadly three types of access control: user-based access control.

(UBAC), role-based access control (RBAC), and attribute-based access control (ABAC). In UBAC, the access control list contains the list of users who are authorized to access data. This is not feasible in clouds where there are many users. In RBAC (introduced by Ferraiolo and Kuhn [10]), users are classified based on their individual roles. Data can be accessed by users who have matching roles. The roles are defined by the system. For example, only faculty members and senior secretaries might have access to data but not the junior secretaries. ABAC is more extended in scope, in which users are given attributes, and the data has attached access policy. Only users with valid set of attributes, satisfying the access policy, can access the data. For instance, in the above example certain records might be accessible by faculty members with more than 10 years of research experience or by senior secretaries with more than 8 years experience. The pros and cons of RBAC and ABAC are discussed in [11]. There has been some work on ABAC in clouds (for example, [12], [13], [14], [15], [16]). All these work use a cryptographic primitive known as attribute-based encryption (ABE). The eXtensible access control markup language [17] has been proposed for ABAC in clouds [18].

An area where access control is widely being used is health care. Clouds are being used to store sensitive information about patients to enable access to medical professionals, hospital staff, researchers, and policy makers. It is important to control the access of data so that only authorized users can access the data. Using ABE, the records are encrypted under some access policy and stored in the cloud. Users are given sets of attributes and corresponding keys. Only when the users have matching set of attributes, can they decrypt the information stored in the cloud.

It is just not enough to store the contents securely in the cloud but it might also be necessary to ensure anonymity of the user. For example, a user would like to store some sensitive information but does not want to be recognized. The user might want to post a comment on an article, but does not want his/her identity to be disclosed. However, the user should be able to prove to the other users that he/ she is a valid user who stored the information without revealing the identity. There are cryptographic protocols like ring signatures [20], mesh signatures [21], group signatures [22], which can be used in these situations. Ring signature is not a feasible option for clouds where there are a large number of users. Group signatures assume the pre-existence of a group which might not be possible in clouds. Mesh signatures do not ensure if the message is from a single user or many users colluding together. For these reasons, a new protocol known as attribute-based signature (ABS) has been applied. ABS was proposed by Maji et al. [23]. In ABS, users have a claim predicate associated with a message. The claim predicate helps to identify the user as an authorized one, without revealing its identity. Other users or the cloud can verify the user and the validity of the message stored. ABS can be combined with ABE to achieve authenticated access control without disclosing the identity of the user to the cloud.

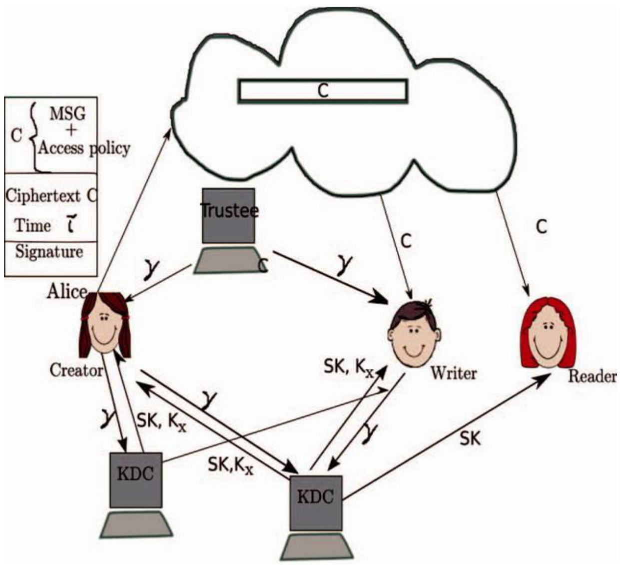
EXISTING SYSTEM

Existing work on access control in cloud are centralized in nature. Except [38] and [18], all other schemes useABE. The scheme in [38] uses a symmetric key approach and does not support authentica-tion. The schemes [12], [13], [16] do not support authentica-tion as well. Earlier work by Zhao et al. [15] provides privacy preserving authenticated access control in cloud. However, the authors take a centralized approach where a single key distribution center (KDC) distributes secret keys and attributes to all users. Unfortunately, a single KDC is not only a single point of failure but difficult to maintain because of the large number of users that are supported in a cloud environment. We, therefore, emphasize that clouds should take a decentralized approach while distributing secret keys and attributes to users. It is also quite natural for clouds to have many KDCs in different locations in the world. Although Yang et al. [34] proposed a decentralized approach, their technique does not authenticate users, who want to remain anonymous while accessing the cloud. In an earlier work, Ruj et al. [16] proposed a distributed access control mechanism in clouds. However, the scheme did not provide user authentication. The other drawback was that a user can create and store a file and other users can only read the file. Write access was not permitted to users other than the creator. In the preliminary version of this paper [1], we extend our previous work with added features that enables to authenticate the validity of the message without revealing the identity of the user who has stored information in the cloud. In this version we also address user revocation that was not addressed in [1]. We use ABS scheme [24] to achieve authenticity and privacy. Unlike [24], our scheme is resistant to replay attacks, in which a user can replace fresh data with stale data from a previous write, even if it no longer has valid claim policy. This is an important property because a user, revoked of its attributes, might no longer be able to write to the cloud. We, therefore, add this extra feature in our scheme and modify [24] appropriately. Our scheme also allows writing multiple times which was not permitted in our earlier work [16].

PROPOSED SYSTEM

We propose our privacy preserving authenticated access control scheme. According to our scheme a user can create a file and store it securely in the cloud. This scheme consists of use of the two protocols ABE and ABS, as discussed in Sections 3.4 and 3.5, respectively. We will first discuss our scheme in details and then provide a concrete example to demonstrate how it works. We refer to the Fig. 1. There are three users, a creator, a reader, and writer. Creator Alice receives a token from the trustee, who is assumed to be honest. A trustee can be someone like the federal government who manages social insurance numbers etc. On presenting her id (like health/social insurance number), the trustee gives her a token . There are multiple KDCs (here 2), which can be scattered. For example, these can be servers in different parts of the world. A creator on presenting the token to one or more KDCs receives keys for encryption/decryption and signing. In the Fig. 1, SKs are secret keys given for decryption, K_x are keys for signing. The message MSG is encrypted under the access policy X. The access policy decides who can access the data stored in the cloud. The creator decides on a claim policy Y, to prove her authenticity and signs the message under this claim. The ciphertext C with signature is c, and is sent to the cloud. The cloud verifies the signature and stores the ciphertext C. When a reader wants to read, the cloud sends C. If the user has attributes matching with access policy, it can decrypt and get back original message.

IMPLEMENTATION



MODULES:

- System Initialization.
- User Registration.
- KDC setup.
- Attribute generation.
- Sign.
- Verify.

MODULES DESCRIPTION:

System Initialization:

Select a prime q, and groups G1 and G2, which are of order q. We define the mapping ˆe : G1 ×G1 → G2. Let g1, g2 be generators of G1 and hj be generators of G2, for j ∈ [tmax], for arbitrary tmax. Let H be a hash function. Let A0 = ha0 0 , where a0 ∈ Z* q is chosen at random. (TSig,TV er) mean TSig is the private key with which a message is signed and TV er is the public key used for verification. The secret key for the trustee is TSK = (a0, TSig) and public key is TPK = (G1,G2,H, g1,A0, h0, h1, . . . , htmax, g2, TV er).

User Registration:

For a user with identity Uu the KDC draws at random Kbase ∈ G. Let K0 = K1/a0 base . The following token γ is output γ = (u,Kbase,K0, ρ), where ρ is signature on u||Kbase using the signing key TSig.

KDC setup:

We emphasize that clouds should take a decentralized approach while distributing secret keys and attributes to users. It is also quite natural for clouds to have many KDCs in different locations in the world. The architecture is decentralized, meaning that there can be several KDCs for key management.

Attribute generation

The token verification algorithm verifies the signature contained in γ using the signature verification key TV er in TPK. This algorithm extracts Kbase from γ using (a, b) from ASK[i] and computes Kx = K1/(a+bx) base , x ∈ J[i, u]. The key Kx can be checked for consistency using algorithm ABS.KeyCheck(TPK,APK[i], γ,Kx), which checks ˆe(Kx,AijBx ij) = ˆe(Kbase, hj), for all x ∈ J[i, u] and j ∈ [tmax].

Sign:

The access policy decides who can access the data stored in the cloud. The creator decides on a claim policy Y, to prove her authenticity and signs the message under this claim. The ciphertext C with signature is c, and is sent to the cloud. The cloud verifies the signature and stores the ciphertext C. When a reader wants to read, the cloud sends C. If the user has attributes matching with access policy, it can decrypt and get back original message.

Verify:

The verification process to the cloud, it relieves the individual users from time consuming verifications. When a reader wants to read some data stored in the cloud, it tries to decrypt it using the secret keys it receives from the KDCs.

CONCLUSION & FUTURE WORK

We have presented a decentralized access control technique with anonymous authentication, which provides user revocation and prevents replay attacks. The cloud does not know the identity of the user who stores information, but only verifies the user's credentials. Key distribution is done in a decentralized way. One limitation is that the cloud knows the access policy for each record stored in the cloud. In future, we would like to hide the attributes and access policy of a user.

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The most expensive operations involving pairings and is done by the cloud. If we compare the computation load of user during read we see that our scheme has comparable costs. Our scheme also compares well with the other authenticated scheme

We present the computation complexity of the privacy preserving access control protocol. We will calculate the computations required by users and that by the cloud. Table 2 presents notations used for different operations.

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INNOVATIVE USES OF SCAP PROTOCOL IN AUTOMATED MALWARE ANALYSIS

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ABSTRACT: ***Malware is increasingly customized and targeted, which means that it has become much harder to detect using conventional methods used generally, like antimalware software. Antimalware software is likely to only locate active infections, not evidence of previous infections. Instead of relying only on antimalware software, organizations can also utilize the SCAP (Security Content Automation Protocol), that is a suite of specifications that standardize the data formats and protocols by which security configuration information and software flaw is communicated, both to machines and humans. SCAP version 1.2 has capabilities to look for various types of system observables evidence such as malware artifacts on Windows hosts. This document illustrates the creation of SCAP content for several Windows malware artifact types and for groups of artifacts that can be shared in responding to an incident.***

Innovative use of Security Content Automation Protocol (SCAP) content in detecting malware increase automation and reduce manual effort to obtain assessment results, with corrective actions needed and provide substantial cost savings. A primary benefit of SCAP content is the ability to make and modify your own checklists.

Index Terms: ***Malware, SCAP, Artifact, Antimalware, Windows hosts.***

I. INTRODUCTION

The word 'Malware' is a Spanish word, combines "mal" that means "bad," and terms as "badware"; also short for "MALalicious softWARE" as easy to remember it. As term implies, malware software programs does damage or does unwanted actions on a host it does exists. Common examples of malwares include Trojan horses, Viruses, Spyware, and Worms. Malware are implemented and deployed at targets with intent of compromising the victim's data for extended period without their knowledge to cause intentional harm.

Over the last three decade, the malware has become an online crime story. Today, estimates of the number of known computer threats such as viruses, worms, backdoors, trojans, exploits, password stealers, spyware, and other variations of unwanted software range into the millions. Ever since criminal malware developers began using client and server polymorphism (the ability for malware to dynamically create different forms of itself to thwart antimalware programs), it has become increasingly difficult to answer the question "How many threat variants are there?" Polymorphism means that there can be as many threat variants as infected computers can produce; that is, the number is only limited by malware's ability to generate new variations of itself.

Anti-malware programs can combat malware in two ways:

1. They can provide real time protection against the installation of malware software on a computer.

2. Anti-malware software programs can be used solely for detection and removal of malware software that has already been installed onto a computer.

A significant evolution has occurred in the malware landscape over the past five years - a change of intent from amateur virus writers seeking attention to professional criminals seeking profit. Today's malware is decidedly different. Instead of hijacking the computer for illicit gains, today's malware is intent on hijacking the user for hard currency, credit card fraud, and outright identity theft. In the current landscape, malware is no longer the end to the means, but rather the means through which the end is reached.

This paper presents the innovative use of Security Content Automation Protocol (SCAP) content in detecting malware in windows family to increase automation and reduce manual effort to obtain assessment results, determine those corrective actions needed and thus provide substantial cost savings.

II. Related Work

Related work is in place emerging to demonstrate how an organization can detect potentially malicious files utilizing OVAL Language Design. The OVAL Language has been created to facilitate the exchange of information between security vendors and tools. The language is composed of three XML schemas; the OVAL Definition schema expresses a specific machine state, the OVAL System Characteristics schema stores configuration information gathered from a system, and the OVAL Results schema presents the output from a comparison of an OVAL Definition against an OVAL System Characteristics instance.

This involves following basic components:

OVAL Definition Schema

<definitions>

<!-- The definitions section serves as the container for each of the definitions contained within an instance document. -->

<tests>

<!-- The tests section contains all of the individual system level tests needed by the definitions. For each platform or application addressed by OVAL, there is a corresponding set of tests to express the various configuration states for that entity. -->

</tests>

<objects>

<!-- The objects section contains all of the system entities (e.g. files, Registry keys, packages…) that are being analyzed by a definition-->

</objects>

<states>

<!-- The states section complements the objects section, in that it contains the values against which the actual system objects are being compared. -->

</states>

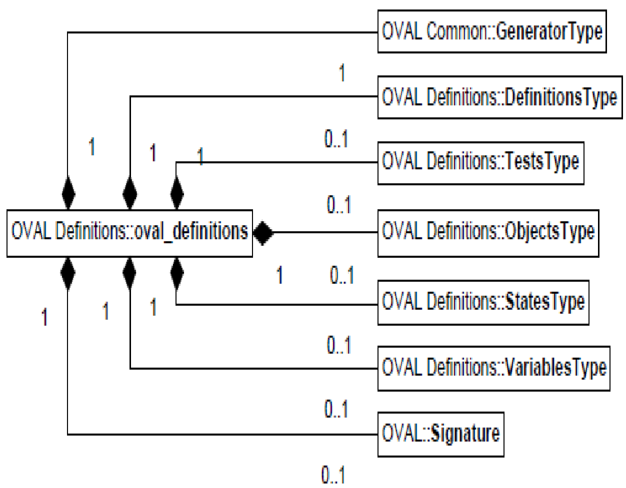
<variables>

<!-- It is not always the case that all of the values within a definition are known at the time a definition is written. Some values can vary based upon an organization's policy or the specific configuration of a machine. -->

</variables>

</definitions>

Typical flowchart can be given as:



OVAL System Characteristics Schema

The <system_info> section contains the details about the installed operating system, system architecture, and network interfaces.

OVAL Results Schema

<results> - The results section serves as a container for the analysis results of potentially

numerous systems against the OVAL Definition document included in the oval_definitions section.

The OVAL Definition schema provides the framework for constructing logical statements of the following form:

Is operating system X installed?

AND Is service Y enabled?

AND Is user Z allowed to use service Y?

Each of these questions is represented by a specific machine configuration parameter, and an associated value. The collection of these questions results in an expression of a desired machine state, which can be applied to a specific system to determine its configuration in comparison. Each of these questions is represented by a specific machine configuration parameter, and an associated value.

III. System Model

Automated Malware Analysis using SCAP Protocol involves typically two ways of detection:

1) The Basics of Malware Detection

Malware is increasingly customized and targeted, which means that it has become much harder to detect using conventional methods, such as antimalware software. Also, antimalware software is likely to only locate active infections, not evidence of previous infections. Because many malware threats largely disappear from a host after a period of time or an event (e.g., memory only malware vanishing after a host reboot), antimalware software may miss them entirely, but the organization still needs to be aware that an infection had occurred, if for no other reason than to mitigate the vulnerability that the malware exploited.

Note that seasoned malware analysts typically possess the specialized skills, training, and experience necessary to review the evidence from a malware infection and determine how additional instances of the same or similar infection on other hosts can be detected. It is unrealistic to expect a novice analyst to be able to perform a triage analysis of malware and determine which artifact or set of artifacts should be used for detection purposes.

2) SCAP and Malware Artifacts Detection

SCAP version 1.2 has capabilities to look for various types of malware artifacts on Windows hosts.

These types include:

- Files (executables, drivers, dynamic link libraries, etc.)
- File hashes
- Registry entries
- Processes
- Services
- Network sockets
- Domain Name System (DNS) cache entries

Note that these artifacts are generally persistent. SCAP version 1.2 is most useful for identifying persistent artifacts; it lacks capabilities to identify more volatile artifacts, such as a malicious executable package that is downloaded, extracted, installed, and then automatically deleted.

There are some significant differences between SCAP malware artifact detection content and other types of SCAP content. The primary difference is that most SCAP existing content is static, such as security configuration baselines for major operating system releases. Such content may not be updated for years at a time, if at all, after its initial release.

Meanwhile, malware artifact detection content may change rapidly—even on a day-to-day basis. This volatility requires content maintainers to diligently monitor reputable threat intelligence sources and proactively develop content to detect new threats, enhance detection capabilities for existing threats, and remove content for threats that are no longer relevant.

IV. Architecture Design

The present version of our architecture is targeting Automating the malware analysis. The scheme in Figure 1 describes the main components of the proposed multi-tier system: Malware artifact, Previous Database collection, SCAP APIs.

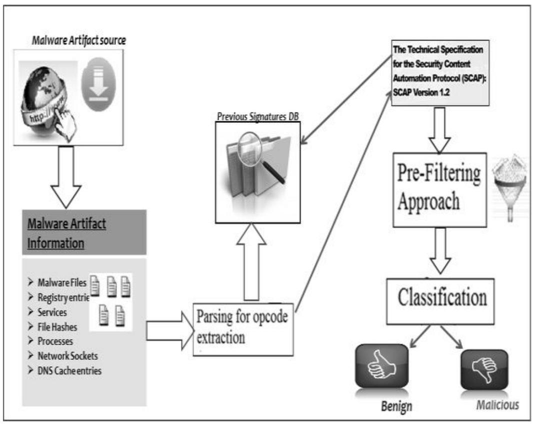


Fig.1: Automated Malware Analysis Architecture Design

The main architecture modules are:

1. Malware Artifact Information (Open/Paid Source):

This module stage is to collect the information about, How did the malware initially get on the system? How does the malware move about and get on other systems? By this information Understanding and applying these malware characteristics enables analysts to understand, detect, locate, and mitigate malware within an infrastructure.

2. SCAP Content module:

In SCAP content module stage, analysts start generating malware detection scripts to detect information available from previous stage. This phase involves all implementation of code for detecting malware based on malware artifact.

3. Malware Detection module

By using this module we can overcome all records of vulnerable system affected with malwares. Since it is all automated no need of human intervention is needed unless it generates the report.

4. Malware Analysis

It defines the behavioral of malwares and its existence in present system. This report should be properly analyzed to check the system criticality to exploit levels and its corresponding effects on system.

Picking up an architecture that can efficiently support an ever-expanding malware data is the first design challenge. The next design challenge is providing ability to known-good readable format. This functionality is critical since the end result of any successful analysis run fully collected information. Solving the challenges outlined above often begins with choosing appropriate system architecture.

SCAP Automated Malware Model Algorithm:

SCAP Automated Malware Model Algorithm (SAMMA) has a simple algorithm which involves step by step procedure to implement a code for detecting malware on a system.

Algorithm: Finding system is malicious or benign

Input: Malware artifact information and target system

Output: Result file indicating target system status and remediation data
1: Get malware_artifact_info
2: choose matching SCAP API module
3: write SCAP OVAL signature
4: scan with Ovaldi interpreter on target system
5: generate results
6: Analyse results
  if result is true
    malicious
  else
    benign
7: remediate using results file

The algorithm states that if there are n malwares then n number of time signature has to be written and need to be run in one time or run in regular time intervals. The implementation of each malware detection code involves every steps of SAMMA algorithms are performed with high quality manner; so that the written signature should not give false positive results.

V. IMPLEMENTATION

The main focus is on developing a better tool to detect the malwares it may be the files, registry entry, file hash, malware services, processes, DNS Cache entries and network Socket taking into consideration both time and space complexity. This section explains how to create SCAP content based on OVAL definitions to identify potential malware in Windows operating system.

A. Detecting Malicious File System Artifacts

OVAL 5.10 provides two test definitions that can be utilized to detect possibly malicious files by matching their name or cryptographic hash to a known bad counterpart:

- `<oval:file_test>`
- `<oval:filehash58_test>`

For reliable malware detection, matching based on a file hash value (generated using a strong, collision-resistant cryptographic hashing algorithm) is preferred over the use of a filename since the hash of a specific binary file is unique to that file. Vastly different binary files could have identical names, so a match on cryptographic hashing for a malicious binary file carries much higher confidence than a similar match on the filename.

B. Detecting Malicious Activity via Flat Log Files

Windows operating system components as well as installed applications generate and write events to text log files regarding activity transpiring on the system. These logs include Windows firewall logs, Windows task scheduler logs, and antimalware software logs. As malware acts upon a system by invoking system processes, launching executables, dropping malicious files, and establishing network connections, log entries may be created in these various log files, allowing these log files to be searched later to identify potential signs of compromise.

OVAL5.10 provides

`<oval:textfilecontent54_test>` definitions that can be utilized to detect malicious activity by matching on specific log entries found on the system within various log files.

C. Detecting Malicious Registry/Service Artifacts

A large portion of the Windows operating system's configuration is stored in data contained in the registry. The registry is a database that contains information about system hardware, installed programs and settings, and user accounts on the system. Programs routinely modify and create keys and values in the registry.

OVAL 5.10 provides the following test definitions to enumerate and compare the properties of registry keys and services:

- `<oval:registry_test>`
- `<oval:service_test>`

When these operations pertain to malware, checking the registry for artifacts can be a useful method of identifying a compromise.

D. Detecting Malicious Process Artifacts

During the lifecycle of an infection it is likely that at least one executable file will be executed and appear in the process list of a victim system for some period of time. OVAL5.10 provides `<oval:process58_test>` definition for enumerating and comparing the properties of running processes.

E. Detecting Malicious Network Related Artifacts

Once the connection between the malware sample and the node has been established, the presence of the malware on the system can be detected by identifying the system's network sockets that have connections to a known bad IP address. Another method is to search through the system's DNS cache for known bad domain names, since the system would need to resolve any domain name to an IP address that could be stored in the DNS cache for later reuse.

OVAL 5.10 supports both methods described above by providing the following test definitions to enumerate and compare the network sockets and DNS cache:

- `<oval:port_test>`
- `<oval:dnscache_test>`

The determination of which OVAL test should be used is made based on which artifacts are available for a given malware sample.

OVAL 5.10 has the ability to combine tests taken together to provide one pass/fail result. These tests are grouped into one `<oval:definition>` in its `<oval:criteria>` element. Every `<oval:criteria>` element represents checks that are grouped together with logical ANDs or ORs. This arrangement allows content creators to specify almost any level of complexity as required arriving at the one meaningful result.

The following use case diagram depicts how the user will interact with system for data with authorized access whereas it also depicts how the malicious user interacts to gain unauthorized access and steal user's data.

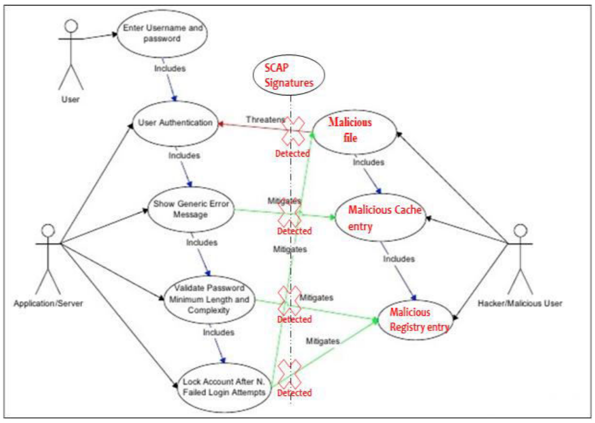


Fig.2: Use Case diagram

The SCAP signatures plays main role detecting such malicious activity and notifies the remedy for machine users on regular bases if user run the SCAP signature or use automation for activity.

Considering above example assume that there are many malicious files in C:\Users folders and its subdirectories the following pseudocode example of an OVAL test can be used for detecting the presence of malicious filenames in the "C:\Users" directory and subdirectories on Windows 7 hosts.

```
<tests>
 <file_test
 xmlns=http://oval.mitre.org/XMLSchema/oval
  definitions-5#windows
 id="oval:edu.mrcet.win7.fname:tst:1"
  version="1" comment="Test to detect malicious
  filenames in the 'Users' directory path and
  subdirectories on Windows 7 hosts."
  check="all">
  <object
 object_ref="oval:edu.mrcet.win7.fname:obj:1"/>
 </file_test>
</tests>

<objects>
 <file_object
 xmlns=http://oval.mitre.org/XMLSchema/oval354
 definitions-5#windows
 id="oval:edu.mrcet.win7.fname:obj:1" version="1"
 comment="Recursively check 'Users' directory
 and subdirectories paths for
```

```
matching
 filenames.">
  <behaviors   max_depth="-1"
  recurse_direction="down"/>
  <path
datatype="string">C:\Users</path
>
  <filename datatype="string"
operation="case
  insensitive equals"

var_ref="oval:edu.mrcet.win7.fna
me:var:1"
  var_check="at least one"/>
 </file_object>
</objects>
<variables>
 <constant_variable

id="oval:edu.mrcet.win7.fname:va
r:1" version="1"
 comment="List of malware
filenames passed to the object
element."
 datatype="string">

<value>malware_filename1.exe</va
lue>

<value>malware_filename2.dll</va
lue>

<value>malware_filename3.dat</va
lue>
 </constant_variable>
</variables>
```

Note that it is sample pseudocode; to run in system it should be included in definition borders and need to interpret with OVAL interpreter. Then by OVAL interpreter will collect system information and match it with signature data to verify. If system data is matches with signature; system will be classified as malicious to that particular malware.

In the above OVAL example the <oval:file_test> element utilizes its child element <oval:file_object> to recursively enumerate through the specified directory and match on the list of filename strings provided to it by the variable element <oval:constant_variable>. In same way we can implement for other SCAP APIs for different malware artifacts too.

VI. Conclusion and Future Scope

The speed, stealth, and purpose of malware are evolving rapidly. Over recent years, substantial technology has emerged to help mitigate risks of fast spreading threats, and a variety of technologies has emerged to begin to help mitigate risks from previously unseen threats. However, malware is becoming both increasingly stealthy, and increasingly malicious in the sense of collection of private and directly valuable personal information. An organization discovers that one of their systems has been compromised by some malicious software. Immediately, the organization tasks their forensics team with investigating the infected systems.

During the investigation, the forensics team notices that the infected system contains certain files that have been modified and that a previously undisclosed vulnerability was used to gain access to the system. Realizing that there is no publicly available check for this vulnerability, the forensics team creates an OVAL Definition with tests to check for the presence of the modified files found during the investigation.

Next Steps:

This report presents examples of representing different malware artifacts on Windows hosts, such as files, file hashes, registry entries, processes, services, network sockets, and DNS entries in SCAP content. The produced SCAP content will be tested thoroughly with an SCAP validated tool to identify the gaps in the SCAP specifications or the tools to detect the presence of malware in an operational environment.

So this research will help organizations understand how to generate the SCAP content in order to leverage their existing SCAP validated tools to detect the presence of malware in an operational environment. Additional research needs to be performed to:

- Identify a user-friendly way to produce the SCAP content
- It should be in more interactive human-readable format and
- An automated mechanism to convert malware artifacts represented in a machine-readable format into an SCAP format.

Collectively these changes will dramatically improve the insight and oversight of the security and integrity of the systems.

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EFFICIENT RE-RANKING FRAMEWORK FOR QUERY-SPECIFIC SEMANTIC SPACES

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Abstract-- **Image re-ranking, as an effective way to improve the results of web-based image search has been adopted by current commercial search engines. A query keyword, a pool of images are first retrieved by the search engine based on textual information. Asking the user to select a query image from the set the minus images are re-ranked based on their visual similarities with the query image. Query and Image based recommendation sorted by the method of re-ranking provides an accurate output of images based on the visual semantic signatures of the query image .In query based recommendation, keyword expansions help provide better results whereas in image recommendation, re-ranking based on priority of images accessed by other users provides more accurate results. At the online stage, images are re-ranked by comparing their semantic signatures obtained from the visual semantic space specified by the query keyword.**

Index Terms: Image search, image re-ranking, semantic space, semantic signature, key word expansion.

INTRODUCTION

The primary objective of this paper is to provide accurate search results based on keyword expansion as well as comparing the semantic signatures of images to provide re-ranked images for the users. The application will feature a search box for typing queries as well as have an option to browse and open the image which the user requires to search for in the web. There are two stages: offline stage and online stage. Semantic signatures of any image queried by the user is calculated and stored in database at the offline stage. Most of the work is done at the offline stage. At the online stage, the user receives re-ranked images those are calculated using semantic signatures at the offline stage. A novel framework is proposed for web image re-ranking. Instead of developing a universal concept dictionary it learns different visual semantic spaces for different query keywords individually and automatically. For example, if the query keyword is "apple", the semantic concepts of "mountains" and "Paris" are unlikely to be relevant and can be ignored. Instead, the semantic concepts of "computers" and "fruit" will be used to learn the visual semantic space related to "apple".

They removed other potentially unlimited number of non-relevant concepts, which serve only as noise and deteriorate the performance of re-ranking in terms of both accuracy and computational cost. The visual features of images are then find into their related visual semantic spaces to get semantic signatures. Web-scale image search engines mostly use keywords as queries and rely on surrounding text to search images.

It is well known that they suffer from the ambiguity of query keywords. For example, using "apple" as query, the retrieved images belong to different categories, such as "red apple", "apple logo", and "apple laptop". Online image re ranking has been shown to be an effective way to improve the image search results. Major internet image search engines have since adopted the re-ranking strategy. Its diagram is shown in Figure 1. Given a query keyword input by a user, according to a stored word-image index file, a pool of images relevant to the query keyword are retrieved by the search engine. By asking a user to select a query image, which reflects the user's search intention, from the pool, the remaining images in the pool are re-ranked based on their visual similarities with the query image. The visual features of images are pre-computed offline and stored by the search engine.

The main online computational cost of image re-ranking is on comparing visual features. In order to achieve high efficiency, the visual feature vectors need to be short and their matching needs to be fast. Another major challenge is that the similarities of low level visual features may not well correlate with images' high-level semantic meanings which interpret users' search intention. To narrow down this semantic gap, for offline image recognition and retrieval, there have been a number of studies to map visual features to a set of predefined concepts or attributes as semantic signature. However, these approaches are only applicable to closed image sets of relatively small sizes. They are not suitable for online web- based image re-ranking. According to our empirical study, images retrieved by 120 query keywords alone include more than 1500 concepts. Therefore, it is difficult and inefficient to design a huge concept dictionary to characterize highly diverse web images.

OUR APPROACH

In this system, a novel framework is proposed for web image re-ranking. Instead of constructing a universal concept dictionary, it learns different visual semantic spaces for different query keywords individually and automatically. We believe that the semantic space related to the images to be re-ranked can be significantly narrowed down by the query keyword provided by the user. For example, if the query keyword is "apple", the semantic concepts of "mountains" and "Paris" are unlikely to be relevant and can be ignored. Instead, the semantic concepts of "computers" and "fruit" will be used to learn the visual semantic space related to "apple".

The query specific visual semantic spaces can more accurately model the images to be re-ranked, since they have removed other potentially unlimited number of non-relevant concepts, which serve only as noise and deteriorate the performance of re-ranking in terms of both accuracy and computational cost. The visual features of images are then projected into their related visual semantic spaces to get semantic signatures. At the online stage, images are re-ranked by comparing their semantic signatures obtained from the visual semantic space of the query keyword.

Our experiments show that the semantic space of a query keyword can be described by just 20, 30 concepts (also referred as "reference classes" in our paper). Therefore the semantic signatures are very short and online image re ranking becomes extremely efficient. Because of the large number of keywords and the dynamic variations of the web, the visual semantic spaces of query keywords need to be automatically learned. Instead of manually defined, under our framework this is done through keyword expansions. Another contribution of the paper is to introduce a large scale benchmark database1 with manually labeled ground truth for the performance evaluation of image re-ranking. It includes 120; 000 labeled images of around 1500 categories (which are defined by semantic concepts) retrieved by the Bing Image Search using 120 query keywords. Experiments on this benchmark database show that 20%35% relative improvement has been achieved on re-ranking precisions with much faster speed by our approach, compared with the state-of-the-art methods.

RELATED WORK

Content-based image retrieval uses visual features to calculate image similarity. Relevance feedback was widely used to learn visual similarity metrics to capture users' search intention. However, it required more users' effort to select multiple relevant and irrelevant image examples and often needs online training. For a web-scale commercial system, users' feedback has to be limited to the minimum with no online training. Cui et al. proposed an image re-ranking approach which limited users' effort to just one-click feedback. Such simple image re-ranking approach has been adopted by popular web-scale image search engines such as Bing and Google recently, as the" find similar images" function.

The key component of image re-ranking is to compute the visual similarities between images. Many image features have been developed in recent years. However, for different query images, low-level visual features that are effective for one image category may not work well for another. To address this, Cui et al. classified the query images into eight predefined intention categories and gave different feature weighting schemes to different types of query images. However, it was difficult for only eight weighting schemes to cover the large diversity of all the web images. It was also likely for a query image to be classified to a wrong category. Recently, for general image recognition and matching, there have been a number of works on using predefined concepts or attributes as image signature. Rasiwasia et al. mapped visual features to a universal concept dictionary.

Lampert et al. used predefined attributes with semantic meanings to detect novel object classes. Some approaches transferred knowledge between object classes by measuring the similarities between novel object classes and known object classes (called reference classes). All these concepts/attributes/reference-classes were universally applied to all the images and their training data was manually selected. They are more suitable for offline databases with lower diversity (such as animal databases and face databases) such that object classes better share similarities. To model all the web images, a huge set of concepts or reference classes are required, which is impractical and ineffective for online image re-ranking.

EXISTING SYSTEM

WEB-SCALE image search engines mostly use keywords as queries and rely on surrounding text to search images. They suffer from the ambiguity of query keywords, because it is hard for users to accurately describe the visual content of target images only using keywords. For example, using "apple" as a query keyword, the retrieved images belong to different categories (also called concepts in this paper), such as "red apple," "apple logo," and "apple laptop."

This is the most common form of text search on the Web. Most search engines do their text query and retrieval using keywords. The keywords based searches they usually provide results from blogs or other discussion boards. The user cannot have a satisfaction with these results due to lack of trusts on blogs etc. low precision and high recall rate. In early search engine that offered disambiguation to search terms. User intention identification plays an important role in the intelligent semantic search engine.

Disadvantages:

* Some popular visual features are in high dimensions and efficiency is not satisfactory if they are directly matched.
* Another major challenge is that, without online training, the similarities of low-level visual features may not well correlate with images' high-level semantic meanings which interpret users' search intention.

PROPOSED SYSTEM

In this paper, a novel framework is proposed for web image re-ranking. Instead of manually defining a universal concept dictionary, it learns different semantic spaces for different query keywords individually and automatically. The semantic space related to the images to be re-ranked can be significantly narrowed down by the query keyword provided by the user. For example, if the query keyword is "apple," the concepts of "mountain" and "Paris" are irrelevant and should be excluded. Instead, the concepts of "computer" and "fruit" will be used as dimensions to learn the semantic space related to "apple."

The query-specific semantic spaces can more accurately model the images to be re-ranked, since they have excluded other potentially unlimited number of irrelevant concepts, which serve only as noise and deteriorate the re-ranking performance on both accuracy and computational cost. The visual and textual features of images are then projected into their related semantic spaces to get semantic signatures. At the online stage, images are re-ranked by comparing their semantic signatures obtained from the semantic space of the query keyword. The semantic correlation between concepts is explored and incorporated when computing the similarity of semantic signatures.

We propose the semantic web based search engine which is also called as Intelligent Semantic Web Search Engines. We use the power of xml meta-tags deployed on the web page to search the queried information. The xml page will be consisted of built-in and user defined tags. Here propose the intelligent semantic web based search engine. We use the power of xml meta-tags deployed on the web page to search the queried information.

The xml page will be consisted of built-in and user defined tags. The metadata information of the pages is extracted from this xml into pdf. our practical results showing that proposed approach taking very less time to answer the queries while providing more accurate information.

Advantages:

- The visual features of images are projected into their related semantic spaces automatically learned through keyword expansions offline.
- Our experiments show that the semantic space of a query keyword can be described by just 20-30 concepts (also referred as "reference classes"). Therefore the semantic signatures are very short and online image re-ranking becomes extremely efficient. Because of the large number of keywords and the dynamic variations of the web, the semantic spaces of query keywords are automatically learned through keyword expansion.

WCHITECTURE

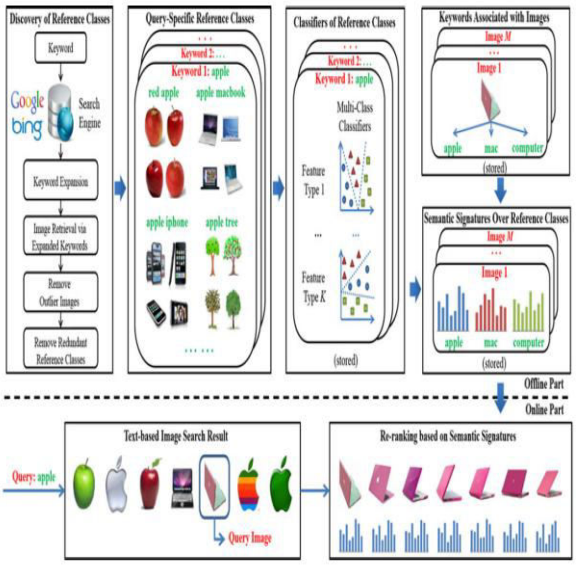

Fig:- Diagram of our new image re-ranking framework.

IMPLEMENTATION

Image Search :-

Many Internet scale image search methods are text-based and are limited by the fact that query keywords cannot describe image content accurately. Content-based image retrieval uses visual features to evaluate image similarity. One of the major challenges of content-based image retrieval is to learn the visual similarities which will reflect the semantic relevance of images. Image similarities can be learned from a large training set where the relevance of pairs of images.

Query Categorization :-

The query categories we considered are: General Object, Object with Simple Background, Scenery Images, Portrait, and People. We use 500 manually labeled images, 100 for each category, to train a C4.5 decision tree for query categorization. The features we used for query categorization are: existence of faces, the number of faces in the image, the percentage of the image frame taken up by the face region, the coordinate of the face center relative to the center of the image.

Visual Query Expansion:-

The goal of visual query expansion is to obtain multiple positive example images to learn a visual similarity metric which is more robust and more specific to the query image. The query keyword is “Paris” and the query image is an image of “eiffel tower”. The image re-ranking result based on visual similarities without visual expansion. And there are many irrelevant images among the top-ranked images. This is because the visual similarity metric learned from one query example image is not robust enough. By adding more positive examples to learn a more robust similarity metric, such irrelevant images can be filtered out. In a traditional way, adding additional positive examples was typically done through relevance feedback, which required more users’ labeling burden. We aim at developing an image re-ranking method which only requires one-click on the query image and thus positive examples have to be obtained automatically.

Images Retrieved by Expanded Keywords :-

considering efficiency, image search engines, such as Bing image search, only re-rank the top N images of the text-based image search result. If the query keywords do not capture the user’s search intention accurately, there are only a small number of relevant images with the same semantic meanings as the query image in the image pool. Visual query expansion and combining it with the query specific visual similarity metric can further improve the performance of image re ranking.

Re-ranking precisions:-

We invited five labelers to manually label testing images under each query keywords into different categories according to their semantic meanings. Image categories were carefully defined by the five labelers through inspecting all the testing images under a query keyword. Each image was labeled by at least three labelers and its label was decided by voting. A small portion of the images are labeled as outliers and not assigned to any category (e.g., some images are irrelevant to the query keywords).Averaged top m precision is used as the evaluation criterion.

Top m precision is defined as the proportion of relevant images among top m re-ranked images. Relevant images are those in the same category as the query image. Averaged top m precision is obtained by averaging top m precision for every query image (excluding outliers). We adopt this criterion instead of the precision-recall curve since in image re-ranking, the users are more concerned about the qualities of top retrieved images instead of number of relevant images returned in the whole

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result set. We compare with two benchmark image re-ranking approaches . They directly compare visual features.

(1) Global Weighting: Predefined fixed weights are adopted to fuse the distances of different low-level visual features.

(2) Adaptive Weighting: proposed adaptive weights for query images to fuse the distances of different low-level visual features. It is adopted by Bing Image Search.

For our new approaches, two different ways of computing semantic signatures as discussed are compared.

a) Query-specific visual semantic space using single signatures (QSVSS Single). For an image, a single semantic signature is computed from one SVM classifier trained by combining all types of visual features.
b) Query-specific visual semantic space using multiple signatures (QSVSS Multiple). For an image, multiple semantic signatures are computed from multiple SVM classifiers, each of which is trained on one type of visual features separately.

CONCLUSION & FUTURE WORK

A unique re-ranking framework is proposed for image search on internet in which only one-click as feedback by user. Specific intention weight schema is used proposed to combine visual features and visual similarities which are adaptive to query image are used. The feedback of humans is reduced by integrating visual and textual similarities which are compared for more efficient image re-ranking. User has only to do one click on image, based on which re-ranking is done. Also duplication of images is detected and removed by comparing hash codes. Image content can be compactly represented in form of hash code. Specific query semantic spaces are used to get more improvised re-ranking of image. Features are projected into semantic spaces which are learned by expansion of keywords.

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A COMPREHENSIVE IN-DEPTH DEFENSIVE SCHEME TO SOLVE KEY-REVELATION CONFLICT

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Abstract: Lately, how to approach the important thing exposure issue in the settings of cloud storage auditing continues to be suggested and studied. Key-exposure resistance happens to be an essential problem for in-depth cyber defense in lots of security applications. To deal with the task, existing solutions all require client to update his secret keys in each and every period of time, which might inevitably generate new local burdens towards the client, especially individuals with limited computation sources, for example cell phones. Within this paradigm, key updates could be securely outsourced with an approved party, and therefore the important thing-update burden around the client is going to be stored minimal. Within this paper, we focus regarding how to result in the key updates as transparent as you possibly can for that client and propose a brand new paradigm known as cloud storage auditing with verifiable outsourcing of key updates. Particularly, we leverage the 3rd party auditor (TPA) in lots of existing public auditing designs; allow it to act as approved party within our situation, making it responsible for both storage auditing and also the secure key updates for key-exposure resistance. Besides, our design also equips the customer with capacity to help verify the validity from the encrypted secret keys supplied by the TPA. Within our design, TPA only must hold an encrypted form of the client's secret key while doing each one of these troublesome tasks with respect to the customer. The customer only must download the encrypted secret key in the TPA when uploading new files to cloud. The safety proof and also the performance simulation reveal that our detailed design instantiations are safe and effective. Each one of these salient features is carefully designed to help make the whole auditing procedure with key exposure resistance as transparent as you possibly can for that client. We formalize the meaning and also the security type of this paradigm.

Keywords: Cloud storage, outsourcing computing, cloud storage auditing, key update, verifiability.

1. INTRODUCTION:

Recently, outsourcing computation has attracted much attention and been researched broadly. Cloud storage is globally viewed among the most significant services of cloud-computing. Although cloud storage provides significant advantage to users, it brings new security challenging problems. An important security problem is how you can efficiently look into the integrity from the data kept in cloud. Recently, many auditing protocols for cloud storage happen to be suggested to cope with this issue [1]. These protocols concentrate on different factors of cloud storage auditing like the high quality, the privacy protection of information, the privacy protection of identities, dynamic data operations, the information discussing, etc. Yu et al. built a cloud storage auditing protocol with key-exposure resilience by updating the user's secret keys periodically. It produces new local burdens for that client since the client needs to execute the important thing update formula in every period of time to create his secret key move ahead. However, it must satisfy several new needs to do this goal. First of all, the actual client's secret keys for cloud storage auditing shouldn't be known through the approved party who performs outsourcing computation for key updates. Otherwise, it'll bring the brand new security threat. Therefore the approved party must only hold an encrypted form of the user's secret key for cloud storage auditing. Next, since the approved party performing outsourcing computation only knows the encrypted secret keys, key updates ought to be completed underneath the encrypted condition. Thirdly, it ought to be extremely powerful for that client to recuperate the actual secret key in the encrypted version that's retrieved in the approved party. Lastly, the customer will be able to verify the validity from the encrypted secret key following the client retrieves it in the approved party. The aim of this paper would be to design a cloud storage auditing protocol that may satisfy above needs to offer the outsourcing of key updates. We advise a brand new paradigm known as cloud storage auditing with verifiable outsourcing of key updates. We design the very first cloud storage auditing protocol with verifiable outsourcing of key updates. We formalize the meaning and also the security type of the cloud storage auditing protocol with verifiable outsourcing of key updates. We prove the safety in our protocol within the formalized security model and justify its performance by concrete implementation [2].

2. SYSTEM MODEL:

You will find three parties within the model: the customer, the cloud and also the third-party auditor (TPA). The customer has the files which are submitted to cloud. The entire size these files isn't fixed, that's, the customer can upload the growing files to cloud in various time points. The cloud stores the client’s files and offers download service for that client [3]. The TPA plays two important roles: the very first is to audit the information files kept in cloud for that client the second reason is to update the encrypted secret keys from the client in every period of time. The TPA can be viewed as like a party with effective computational capacity or perhaps a service in another independent cloud. Within the finish of every period of time, the TPA updates the encrypted client’s secret key for cloud storage auditing based on the next time period [4]. We use three games (Game 1, Game 2 and Game 3) to explain the adversaries with various compromising abilities who're from the security from the suggested protocol. Game 1 describes a foe, which fully compromises the TPA to obtain all encrypted secret keys. Game 2 describes a foe, which compromises the customer to obtain DK, attempts to forge a legitimate authenticator in almost any period of time. Game 3 offers the foe more abilities, which describes a foe, which compromises the customer and also the TPA to obtain both ESKj and DK previously period j, attempts to forge a legitimate authenticator before period of time j. The safety model formalizes the adversaries with various reasonable abilities who attempt to cheat the challenger he owns one file he actually doesn't entirely know.

3. PROPOSED MODEL:

We make use of the same binary tree structure to evolve keys that has been accustomed to design several cryptographic schemes. This tree structure could make the protocol achieve fast key updates and short key size. One problem we have to resolve would be that the TPA should carry out the outsourcing computations for key updates underneath the condition the TPA doesn't be aware of real secret key from the client. Traditional file encryption strategy is not appropriate since it helps make the key update hard to be completed underneath the encrypted condition. Besides, it will likely be even more complicated to allow the customer using the verification capacity to guarantee the validity from the encrypted secret keys. To deal with these challenges, we advise look around the blinding technique with homomorphic property to efficiently “encrypt” the key keys. Our security analysis afterwards implies that such blinding technique with homomorphic property can sufficiently prevent adversaries from forging any authenticator of valid messages. Therefore, it will help to make sure our design goal the key updates is as transparent as you possibly can for that client [5]. To Get Rid of the Encrypted Secret Key Verification from the Client, when the client isn't in urgent have to know if the encrypted secret keys downloaded in the TPA are correct, we are able to remove his verifying operations making the cloud carry out the verification operations later. Within this situation, we are able to delete the VerEKey formula from your protocol. Whether it holds, then your encrypted secret key should be correct. In this manner, the customer doesn't need to verify the encrypted secret keys immediately after he downloads it in the TPA. We assess the performance from the suggested plan through several experiments which are implemented with the aid of the Pairing-Based Cryptography (PBC) library. Within the suggested plan, the important thing update workload is outsourced towards the TPA. In comparison, the customer needs to update the key alone in every period of time in plan. Within the designed Sys Setup formula, the TPA only holds a preliminary encrypted secret key and also the client holds an understanding key which is often used to decrypt the encrypted secret key. Within the designed Key Update formula, homomorphic property helps make the secret key capable of being updated under encrypted condition and makes verifying the encrypted secret key possible. The Ver ESK formula could make the customer look into the validity from the encrypted secret keys immediately. We compare the important thing update time on client side between your both schemes. Once the client really wants to upload new files towards the cloud, it must verify the validity from the encrypted secret key in the TPA and recover the actual secret key [6]. We demonstrate time from the challenge generation process, the proof generation process, and also the proof verification process with various quantities of checked data blocks. Within our plan, the communicational messages comprise the task message and also the proof message. Once the client really wants to upload new files towards the cloud, it must verify the validity from the encrypted secret key in the TPA and recover the actual secret key. We show time of these two processes happened in various periods of time. Used, these processes don't take place in the majority of periods of time. They merely take place in time periods once the client must upload new files towards the cloud. In addition, the job for verifying the correctness from the encrypted secret key can fully be carried out by the cloud.

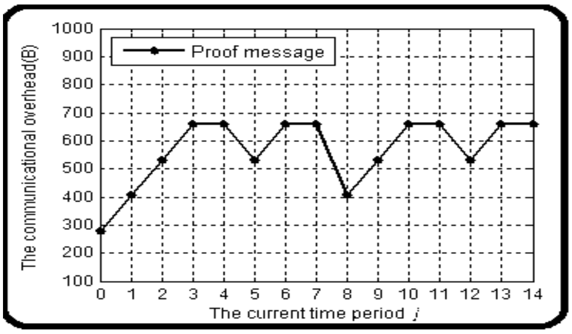

Fig.1.Computational costs

4. CONCLUSION:

Within our design, TPA only must hold an encrypted form of the client's secret key while doing each one of these troublesome tasks with respect to the customer. The customer only must download the encrypted secret key in the TPA when uploading new files to cloud. Within this paper, we study regarding how to delegate key updates for cloud storage auditing with key-exposure resilience. One problem we have to resolve would be that the TPA should carry out the outsourcing computations for key updates underneath the condition the TPA doesn't be aware of real secret key from the client. We demonstrate time from the challenge generation process, the proof generation process, and also the proof verification process with various quantities of checked data blocks. Within our plan, the communicational messages comprise the task message and also the proof message. We advise the very first cloud storage auditing protocol with verifiable outsourcing of key updates. Within this protocol, key updates are outsourced towards the TPA and therefore are transparent for that client. We provide the formal security proof and also the performance simulation from the suggested plan. Additionally, the TPA only sees the encrypted form of the client's secret key, as the client can further verify the validity from the encrypted secret keys when installing them in the TPA.

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PRIVACY GRID SYSTEM FOR LOCATION BASED SERVICES

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Abstract: Location-based services (LBS) require users to continuously report their location to a potentially untrusted server to obtain services based on their location, which can expose them to privacy risks. Unfortunately, existing privacy-preserving techniques for LBS have several limitations, such as requiring a fully-trusted third party, offering limited privacy guarantees and incurring high communication overhead. In this paper, we propose a user-defined privacy grid system called dynamic grid system (DGS); the first holistic system that fulfills four essential requirements for privacy-preserving snapshot and continuous LBS. (1)

The system only requires a semi-trusted third party, responsible for carrying out simple matching operations correctly. This semi-trusted third party does not have any information about a user's location. (2) Secure snapshot and continuous location privacy is guaranteed under our defined adversary models. (3) The communication cost for the user does not depend on the user's desired privacy level, it only depends on the number of relevant points of interest in the vicinity of the user. (4) Although we only focus on range and k-nearest-neighbor queries in this work, our system can be easily extended to support other spatial queries without changing the algorithms run by the semi-trusted third party and the database server, provided the required search area of a spatial query can be abstracted into spatial regions. Experimental results show that our DGS is more efficient than the state-of-the-art privacy-preserving technique for continuous LBS.

Keywords: Dynamic grid systems, location privacy, location-based services, spatio-temporal query processing, cryptography.

INTRODUCTION

In today's world of mobility and ever-present Internet conn activity, an increasing number of people use location-based services (LBS) to request information relevant to their current locations from a variety of service providers. This can be the search for nearby points of interest (POIs) (e.g., restaurants and hotels), location aware advertising by companies, traffic information tailored to the highway and direction a user is traveling and so forth. The use of LBS, however, can reveal much more about a person to potentially untrustworthy service providers than many people would be willing to disclose. By tracking the requests of a person it is possible to build a movement profile which can reveal information about **a user's work (office location), medical records**

(visit to specialist clinics), political views (attending political events), etc. Nevertheless, LBS can be very valuable and as such users should be able to make use of them without having to give up their location privacy. A number of approaches have recently been proposed for preserving the user location privacy in LBS.

In general, these approaches can be classified into two main categories. (1) Fully-trusted third party (TTP). The most popular privacy-preserving techniques require a TTP to be placed between the user and the service **provider to hide the user's** location information from the service provider. The main task of the third party is keeping track of the exact location of all users and blurring a querying **user's location into a cloaked** area that includes **k−1** other users to achieve k-anonymity. This TTP model has three drawbacks. (a) All users have to continuously report their exact location to the third party, even though they do not subscribe to any LBS. (b) As the third party knows the exact location of every user, it becomes an attractive target for attackers. (c) The k-anonymity-based techniques only achieve low regional location privacy because cloaking a region to include k users in practice usually results in small cloaking areas. (2) Private information retrieval (PIR) or oblivious transfer (OT).Although PIR or OT techniques do not require a third party, they incur a much higher communication overhead between the user and the service provider, requiring the transmission of much more information than the user actually needs. Only a few privacy-preserving techniques have been proposed for continuous LBS. These techniques rely on a TTP to continuously expand a cloaked area to include the initially assigned k users. These techniques not only inherit the drawbacks of the TTP model, but they also have other limitations. (1) Inefficiency. Continuously expanding cloaked areas substantially increases the query processing overhead. (2) Privacy leakage. Since the database server receives a set of consecutive cloaked areas of a user at different timestamps, the correlation among the cloaked areas would provide useful information for inferring the **user's location. (3)** Service termination. A user has to terminate the service when users initially assigned to her cloaked are a leave the system.

In this paper, we propose a user-defined privacy grid system called dynamic grid system (DGS) to provide privacy-preserving snapshot and continuous LBS. The main idea is to place a semi-trusted third party, termed query server (QS), between the user and the service provider (SP). QS Only needs to be semi-trusted because it will not collect/store or even have access to any user location information. Semi-trusted in this context means that while QS will try to determine the location of a user, it still correctly carries out the simple matching operations required in the protocol, i.e., it does not modify or drop messages or create new messages. An untrusted QS would arbitrarily modify and drop messages as well as inject fake messages, which is why our system depends on a semi-trusted QS. The main idea of our DGS. In DGS, a querying user first determines a query area, where the user is comfortable to reveal the fact that she is somewhere within this query area. The query area is divided into equal-sized grid cells based on the dynamic grid structure specified by the user. Then, the user encrypts a query that includes the information of the query area and the dynamic grid structure, and encrypts the identity of each grid cell intersecting the required search area of the spatial query to produce a set of encrypted identifiers. Next, the user sends a request including (1) the encrypted query and (2) the encrypted identifiers to QS, which is a semi-trusted party located between the user and SP. QS stores the encrypted identifiers and forwards the encrypted query to SP specified by the user. SP decrypts the query and selects the POIs within the query area from its data base. For each selected POI, SP encrypts its information, using the dynamic grid structure specified by the user to find a grid cell covering the POI, and encrypts the cell identity to produce the encrypted identifier for that POI. The encrypted POIs with their corresponding encrypted identifiers are returned to QS. QS Stores the set of encrypted POIs and only returns to the user a subset of encrypted POIs whose corresponding identifiers match any one of the encrypted identifiers initially sent by the user. After the user receives the encrypted POIs, she decrypts them to get their exact locations and computes a query answer.

PROPOSED SYSTEM

Service providers

Our system supports any number of independent service providers. Each SP is a spatial database management system that stores the location information of a particular type of *static* POIs, e.g., restaurants or hotels, or the store location information of a particular company, e.g., Starbucks or **McDonald's. The spatial database uses an existing** spatial index (e.g., R-tree or grid structure) to index POIs and answer range queries (i.e., retrieve the POIs located in a certain area). As depicted in our system architecture, SP does not communicate with mobile users directly, but it provides services for them indirectly through the query server (QS).

Mobile users

Each mobile user is equipped with a GPS-enabled

device that determines the user's location in the form.

The user can obtain snapshot or continuous LBS from our system by issuing a spatial query to a particular SP through QS. Our system helps the user select a query area for the spatial query, such that the user is willing to reveal to SP the fact that the user is located in the given area. Then, a grid structure is created and is embedded inside an encrypted query that is forwarded to SP, it will not reveal any information about the query area to QS itself. In addition, the communication cost for the user in DGS does not depend on the query area size. This is one of the key features that distinguishes DGS from the existing techniques based on the fully-trusted third party model. When specifying the query area for a query, the user will typically consider several factors.

(1) The user specifies a minimum privacy level, e.g., city level. For a snapshot spatial query, the query area would be the minimum bounding rectangle of the city in which the user is located. If better privacy is required, the user can choose the state level as the minimum privacy level (or even larger, if desired). The size of the query area has no performance implications whatsoever on the user, and a user can freely choose the query area to suit her own privacy requirements. For continuous spatial queries, the user again first chooses a query area representing the minimum privacy level required, but also takes into account possible movement within the time period t for the query. If movement at the maximum legal speed could lead the user outside of the minimum privacy level query area within the query time t, the user enlarges the query area correspondingly. This enlargement can be made generously, as a larger query area does not make the query more expensive for the user, neither in terms of communication nor computational cost. (2) The user can also generate a query area using a desired k-anonymity level as a guideline.

Query servers

QS is a semi-trusted party placed between the mobile user and SP. Similar to the most popular infrastructure in existing privacy-preserving techniques for LBS, QS can be maintained by a telecom operator. The control/data flows of our DGS are as follows: 1) The mobile user sends a request that includes (a) the identity of a user-specified SP, an *encrypted query* (which includes information about the user-defined dynamic grid structure), and (b) a set of *encrypted identifiers* (which are calculated based on the user-defined dynamic grid structure) to QS. 2) QS stores the encrypted identifiers and forwards the encrypted query to the user-specified SP. 3) SP decrypts the query and finds a proper set of POIs from its database. It then encrypts the POIs and their corresponding identifiers based on the dynamic grid structure specified by the user and sends them to QS. 4) QS returns to the user every encrypted POI whose encrypted identifier matches one of the encrypted identifiers initially sent by the user. The user decrypts the received POIs to construct a candidate answer set, and then performs a simple filtering process to prune false positives to compute an exact query answer.

Supported spatial queries

DGS supports the two most popular spatial queries, i.e., range and k-NN queries, while preserving the **user's location privacy. The mobile user registers a** continuous range query with our system to keep track of the POIs within a user-specified distance, *Range*, **of the user's current location for a certain time** period, e.g., "*Continuously send me the restaurants within one mile of my current location for the next one hour*". **The mobile user can also issue a** continuous k-NN query to find the k-nearest POIs to **the user's current location for a specific time period, e.g.,** "*Continuously send me the five nearest restaurants to my current location for the next 30 minutes*". **Since a snapshot query is just the initial** answer of the continuous one, DGS also supports snapshot range and k-NN queries. Although we only focus on range and k-NN queries in this work, DGS is applicable to other continuous spatial queries if the query answer can be abstracted into spatial regions. For example, our system can be extended to support reverse-NN queries and density queries because recent research efforts have shown that the answer of these queries can be maintained by monitoring a region. Our DGS has two main phases for privacy-preserving continuous range query processing. The first phase finds an initial (or a snapshot) answer for a range query, and the second phase incrementally **maintains the query answer based on the user's** location updates User Anonymity De-anonymize a user by using the information contained in the protocol (although they still faithfully follow the protocol itself). While QS does not have any information about a user that would allow it to narrow down the list of users that would fit a specific query, SP has access to the plaintext query of a user. This query, however, only contains the query region and the grid parameters, and with the information available, QS can therefore do no better than establish that the user is somewhere within the query region.

CONCLUSION

In this paper, we proposed a dynamic grid system (DGS) for providing privacy-preserving continuous LBS. Our DGS includes the query server (QS) and the service provider (SP), and cryptographic functions to divide the whole query processing task into two parts that are performed separately by QS and SP. DGS does not require any fully-trusted third party (TTP); instead, we require only the much weaker assumption of no collusion between QS and SP. This separation also moves the data transfer load away from the user to the inexpensive and high-bandwidth link between QS and SP. We also designed efficient protocols for our DGS to support both continuous k-nearest-neighbor (NN) and range queries. To evaluate the performance of DGS, we compare it to the state-of-the-art technique requiring a TTP. DGS provides better privacy guarantees than the TTP scheme, and the experimental results show that DGS is an order of magnitude more efficient than the TTP scheme, in terms of communication cost. In terms of computation cost, DGS also always outperforms the TTP scheme for NN queries; it is comparable or slightly more expensive than the TTP scheme for range queries.

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A SECURE SYSTEM TO SCREEN UNWANTED CONTENT FROM SOCIAL NETWORKS

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Abstract— **Day to day the usage of Social Networks is rapidly increasing but the Social network providers are unable to control the unwanted content like messages/ popup's. So it is very much necessary to give users the ability to control the content posted on their own private space and to avoid the display of unwanted messages/ popup's. In the existing system the social networks provide support to a basic level. In this paper, we propose a secure system for the users of social networks to screen unwanted contents on their respective walls. We utilized the Machine Learning (ML) text categorization techniques to automatically assign with each short text message a set of categories based on its content.**

Keywords— Online Social Networks; Filtered Wall; Radial Basis Function Networks; Short Text Classification;

I. INTRODUCTION

Now a days the considerable amount of interaction between many humans is through Online Social Networks (OSNs). These are the web-based services which allow the users to create their own public profile, a list of users of their wish such that they can share connections. Thus Online Social Networks gave users an ability to share ideas, activities, and interests with people in their network. The communication between the users may be of posting information, images, messages, comments, etc. In fact there is a chance to post or comment on any public/private areas, called general walls. There may be unwanted content on user's private space hence the users should have an ability to control their unwanted content by detecting them. The unwanted content may be vulgar or political etc. However, the basic OSNs have a little bit support to avoid the unwanted content. By information filtering method the users have an ability to control the messages present on respective walls by filtering out the unwanted content. But there is no proper technique to filter the unwanted content.

Up to now, there are no content based preferences to prevent the undesired messages. The support provided by OSNs to prevent this undesired messages has much limitations such as they cannot detect short texts, but the messages generated on the wall are short texts. Therefore we propose a secure automated system called Filtered wall, which can be effectively able to detect the unwanted/undesired messages on respective OSNs user walls. We have targeted the short text messages and hence we adopted the Machine Learning (ML) text categorization techniques. By concentrating on extraction and selection of set of characterizing features we have built a strong short text classifier (STC). For short text classifications we have used neural learning based on Radial Basis Function Networks (RBFN) because they have an ability to manage noisy data and vague classes. Thus we have a flexible language for to specify Filtering Rules (FRs) such that user can specify what to display the message on his/her respective walls i.e. FRs gives user an ability to customize different filtering criteria. And a user can also prevent a list of members to post any kind of message on his wall which is called as BlackLists (BLs).

II. METHODOLOGY

The primary goal of this secure system is to automatically filter the undesired messages from OSNs user walls on basis of the user.

A. *Construction of a Filtered Wall*

Basically the architecture of an OSN is three-tier architecture. The first layer is called Social Network Manager (SNM), which provides the user functionalities of an OSN, second layer is Social Network Applications (SNAs), which is used as a support for SNAs and the third layer is nothing but a user interface of the second layer called Graphical User Interfaces (GUIs).

Because the user interaction with the system is at Graphical User Interfaces (GUIs) in this construction the proposed system is placed in the second and third layers such that the filtered walls and black lists are provided to the user in GUIs. Initially the user enters into the private wall of other user and tries to post a message and the decision to post or screen is done by the user of the respective wall. The process is if user posts a message on others wall then it is intercepted by the filtered wall (FW). Then to extract the metadata a ML-based text classifier is adopted and with the metadata the FW decides to post or to filter the message.

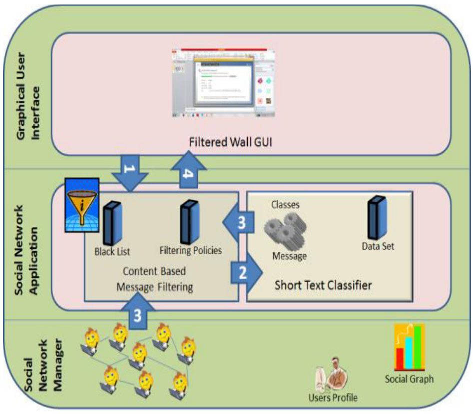

Fig.1: Filtered Wall Conceptual Architecture.

The proposed secure system totally rely on Content-Based Messages Filtering (CBMF) and the Short Text Classifier modules.

Firstly, the classification of messages into set of a set of categories takes place. The component make use of the message categories provided by the STC and applies it to FRs given by the user to his/her wall. For amelioration of this filtering process the BLs are used.

B. *Short Text Classifier*

The techniques established previously worked well for text classifications on large datasets of documents such as newspapers but failed to do in short texts like messages. The critical features are the description of a set of characterizing and discriminant features allowing the representation of underlying concepts and the collection of a complete and consistent set of supervised examples. We had adopted neural learning strategy to classify short text for designing and evaluating previous techniques. The first-level task is conceived as a hard classification in which short texts are labeled with crisp Neutral and Non neutral labels. [8] The second-level soft classifier acts on the crisp set of non-neutral short texts.

The extraction of an appropriate set of features by which representing the text of a given document is a crucial task strongly affecting the performance of the overall classification strategy. In the BoW representation, terms are identified with words. According to Vector Space Model (VSM) for text representation, a text document dj is represented as a vector of binary or real weights dj = w1j, . . . , w|T|j, where T indicates the set of terms that occur at least once in at least one document of the collection Tr, and wkj € [0; 1] denotes how much term tk contributes to the semantics of document dj. In the BoW representation, terms are identified with words. In the case of non-binary weighting, the weight wkj of term tk in document dj is computed according to the standard term frequency—inverse document frequency (tf-idf) weighting function [3], defined as

$$tf-idf(t_k, d_j) = \#(t_k, d_j) \cdot \log \frac{|\mathcal{T}_r|}{\#\mathcal{T}_r(t_k)},$$

where #(tk , dj) denotes the number of times tk occurs in dj , and #Tr (tk) denotes the document frequency of term tk , i.e., the number of documents in Tr in which tk occurs. Domain specific criteria are adopted in choosing an additional set of features, Dp, concerning orthography, known words and statistical properties of messages. Dp features are heuristically accessed. For ex, Correct Words: it expresses the amount of terms tk ∈ T^K, where tk is a term of the considered document dj and K is a set of known words for the domain language. This value is normalized by Σ|T| k=1 #(tk,dj).

And similarly for bad words, they are computed similarly to the Correct words feature, where the set K is a collection of dirty words for the domain language. CFs are not very dissimilar from BoW features describing the nature of data. Therefore, all the formal definitions introduced for the BoW features also apply to CFs.

C. *Classification based on Machine Learning*

Short text categorization is a hierarchical two-level classification process. The first-level classifier does a binary hard classification that labels messages as Neutral and Non-Neutral. The first-level filtering task enables the subsequent second-level task in which a finer-grained classification is performed. The second-level classifier carries out a soft-partition of Non-neutral messages assigning a given message a gradual membership to each of the non-neutral classes.

D. *Filtering Rules to be determined*

Filtering Rules provides an ability to the user to control the unwanted content on his/her respective wall, these are customized by the user only. Hence if a message s posted then it is said to be no in BL. These FRs allow the user to make their own constraints on their respective walls. Given the social network scenario, creators may also be identified by exploiting information on their social graph.

E. *For FRs thresholds an Online Setup Assistant*

Online Setup Assistant (OSA) presents the user with a set of messages selected from the data set. For each message, the user tells the system the decision to accept or reject the message. The collection and processing of user decisions on an adequate set of messages distributed over all the classes allows to compute customized thresholds representing the user attitude in accepting or rejecting certain contents.

F. *Blacklists Rules specification*

BLs are directly managed by the system, and should be able to determine the users to be inserted in the BL and user should give a retention time such that how long the blocked users have been retender. The set of rules to specify these kind of things are called Blacklists Rules.[5] To enhance flexibility such that to avoid messages from undesired creators BL rules are very much important.

The BL mechanism mainly does three things they are, detect the users to be inserted in Black List, blocking of all their messages on respective user specified walls and decide when user retention in the BL is finished.

III. RESULTS AND DISCUSSIONS

Initially the system is provide as an OSN and the user have to register to it and then login into it to connect with other users. The typical look of a login page of an OSN is shown below.

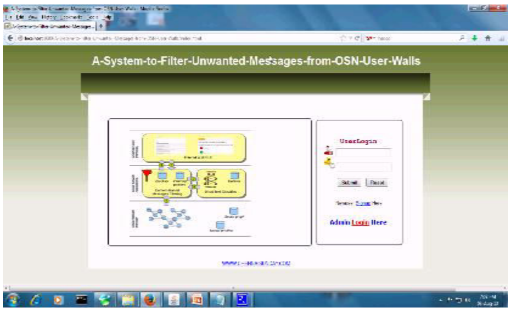

Fig.2: Login page of OSN of proposed system.

After providing correct username and password combo in the login page the user is directed to his personal account i.e. his private space nothing but his wall he can post anything like comment, images, messages etc. A picture of user messaging in on the wall is shown below.

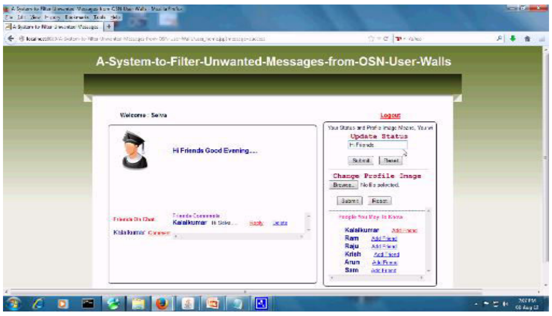

Fig.3: User posting a message.

The step after posting a message is to be verified through FW such that it decides by its methodology and posts it or filters it based on BLs. Here we can see a message posted.

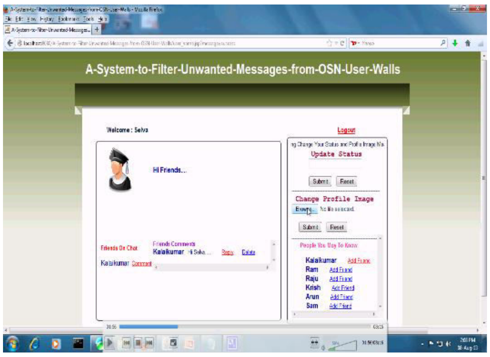

Fig.4: Updating status also follows FR.

And also user gets all the functionalities of a basic OSN such as updating profile picture, accepting and declining requests of respective user. Here the updating of an users profile picture is shown below.

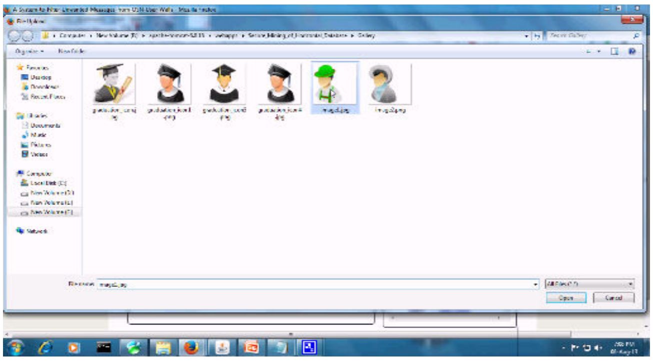

Fig.5: Selection of Profile Picture

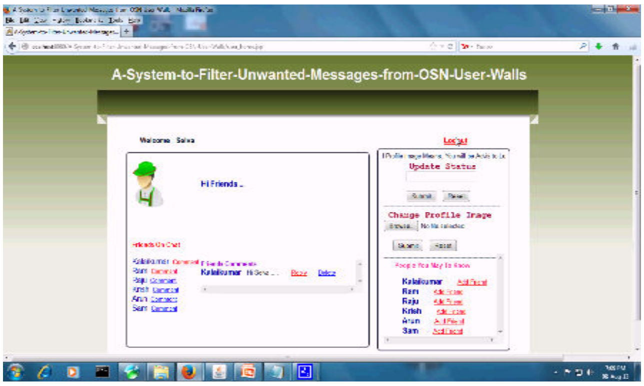

Fig.6: Updated user Filtered Wall

.IV. CONCLUSIONS

In this paper, we have proposed a secure system to the users of OSNs to screen/filter the unwanted content from their respective walls. The customized content based Filter Rules are applied for ease of use. By placing user defined Black Lists the flexibility of the system is greatly increased. Using this system the user gets an ability to control the unwanted messages hence a secure way of protecting walls from unwanted content is determined.

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A NOVEL APPROACH FOR ADVANCED PERSISTENT THREAT DETECTION USING BIG DATA ANALYTICS WITH HADOOP AS SERVICE ANOMALY

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Abstract: **Now a day's cyber-targeted attacks such as Advanced Persistent Threats (APT's) are rapidly growing as a social and national threat. Traditional attacks aim to damage the network or leak the personal information. An Advanced Persistent Threat (APT) is a targeted attack against a high-value asset or a physical system.. Many models are proposed to detect and prevent unwanted attacks. These APT malware Infections can be detected Based on Malicious DNS and traffic Analysis. The current industry approach, which is focused on real-time detection with emphasis on signaturematching, althougheffective against traditional attacks is unable to address the unique characteristics of APT's. The solution to this problem can overcome by Big Data Analytics with Hadoop as Service Anomaly. We want to look at the agents who are carrying particular tasks, by understanding what is happening early on in an attack and on a continuing basis, evaluate the degree of risk at any one time, and have a plan to provide countermeasures more effectively to combat the APTs using Big data Analytics.**

Keywords:* *Big Data Analytics, APT, Malware Infections

I. INTRODUCTION

Advanced Persistent Threat (APT) attacks are increasing on the internet nowadays; unfortunately, they are hard to detect. It is a set of stealthy and continuous hacking processes targeting a specific entity with high-value information, such as government, military and the financial industry. The intention of an APT attack is to steal data rather than to cause damage to the network or organization. Once hacking into the network has been achieved, the attacker would install APT malware on the infected machine. APT malware, for instance, Trojan horse or backdoor, is tailored for anti-virus software and Firewalls of the target network. It is not only used for remotely controlling the compromised machines in the APT attack, but also for stealing sensitive information from infected host over an extended period of time APT malware can evade anti-virus software using polymorphic code, and bypass Firewall using protocol on allowed ports.

DNS is an important component of the Internet, and it is the protocol that is responsible for resolving a domain name [1] to the corresponding IP address. Besides being popular for benign use, such as helping to locate web servers and mailing hosts, domain names are also susceptible to malicious use. APT malware is very different from the bots and worms [2]. The primary purpose of APT malware is to remotely control the machines and to steal confidential data, rather than to launch denial-of-service attacks, send spam emails or cause damage. It requires a high degree of stealth over a prolonged duration of operation. To identify malicious domains that are involved in APT malware activity is a challenge. The crafted malware in APT attack do not use malicious flux service or DGA domains. The domains for APT malware were registered by the attackers. Compared with these bots and worms, crafted malware requires high degree of stealth. For this reason, the DNS behavioral features of APT malware are unconspicuous. It is too hard to analyze large volumes of inbound and outbound traffic in a large network, such as a large enterprise and an ISP. To detect APT malware infections in a large network is another challenging problem.

II. INTRUSION DETECTION STUDIES

In general, the main studies of network intrusion detection include signature-based detection and anomaly-based detection. Signature-based detection is a technology that relies on a existing signature database to detect known malware infections. By using signature-based detection technology, it can identify malware C&C communication traffic through signature-based pattern matching. So for malware infection detection, it is a typical approach. But signature-based detection technology has a fatal drawback, it cannot detect new malware infections if the signature of the new malware is not in the existing signature database.

Anomaly-based intrusion detection [3], [4] is a technology that detect abnormal behaviors that deviates from ``normal'' behaviors. The ``normal'' behaviors of the network need to be studied and identified at first. The primary advantage of anomaly-based intrusion detection is the capability to detect new or unknown attacks. Because the new or unknown malware whose signature is not available would also generate abnormal behaviors. The primary drawback of anomaly-based intrusion detection is that, it is more prone to generating false positives. Because the behaviors of different networks and applications are so complicated, the ``normal'' behaviors is very hard to accurately identify. Different from signature-based detection, anomaly-based intrusion detection is a broader match which is based on detecting abnormal behaviors. Many legitimate applications perform the same abnormal behaviors as malicious ones.

The main limitation is the fact IDnS is not good at detecting malware infections that do not rely on domains, such as the trojan use the IP address directly to locate the command and control server [5].

The solution to this problem can overcome by Big Data Analytics with Hadoop as Service Anomaly.

Big Data Analytics accepts the huge data sets and

Varied data types, both semi-structured and unstructured like videos, images, audio, web-pages, texts or e-mails etc. and convert it into reliable information. Big data analytics describes the simple algorithm for large amount of data without compromising performance. Analysis algorithms are provided directly to database which go beyond the pack and innovate newer more sophisticated statistical analysis. Big Data Analytics use number of tools to do the analysis and processing of data in meaningful way. Hadoop is one of the tools which is aimed to improve the performance of data processing.

III. BIG DATA ANALYTICS

Big data is now a big problem as the volume; variety and velocity of data coming into enterprises continue to reach extraordinary levels. This unexpected growth means that not only must you understand big data in order to process the information that truly counts, but you also must analyze the possibilities of what you can do with big data using big data analytics. Big data analytics is the process of analyzing big data to find hidden patterns, unknown correlations and other useful information that can be extracted to make better decisions. Fig. 1 demonstrates the potential use cases for Big Data Analytics. It shows the relation between data variety i.e. unstructured, semi structured and structured data with data velocity from batch system to real time system. It shows that what different analysis can be done by using Big Data Analytics and at which level.

With big data analytics, scientists and others can analyze huge volumes of data that old analytics and business intelligence solutions can't find. Consider this it's possible that enterprise could accumulate billions of rows of data with hundreds of millions of data combinations in multiple data stores and abundant formats. Fig. 2 is demonstrating the value of Big Data Analytics by drawing the graph between time and cumulative cash flow. Old analytics techniques like any data warehousing application, you have to wait hours or days to get information as compared to Big Data Analytics.

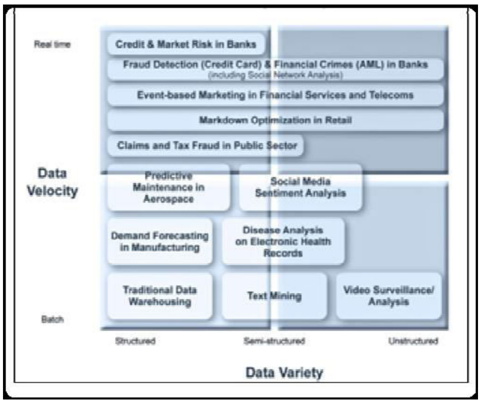

Fig.1. Potential use cases for Big Data Analytics

Information has the timeliness value when it is processed at right time otherwise it would be of no use. It might not return its value at proper cost.

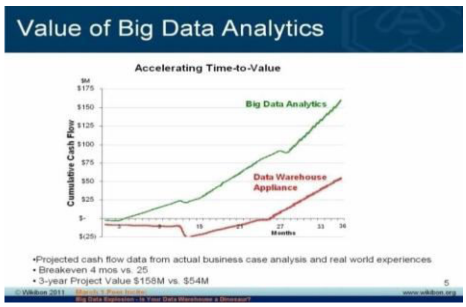

Fig.2. Value of Big Data Analytics

Big data Analytics used the tool Hadoop for processing the unstructured data. The main point is whether Hadoop will become as indispensable as database management systems. Hadoop has proven its advantages of use and cost where volume and variety are extreme [6]. Cloud era, Horton works, and MapR are doing work for Hadoop on high-scale storage and Map Reduce processing data into the world of analytics. Data analysis is a do-or-die requirement for today's businesses. Hadoop upstarts to traditional database players by analysis done by vendors.

IV. HADOOP – TOOL FOR ANALYSIS

Hadoop is a software framework for storing and processing Big Data and work under Big Data Analytics. It is an open-source framework build on java platform and aimed at to improve the performance in terms of data processing on Big Data.

Features of Hadoop:

1. Hadoop has two core components: HDFS and MapReduce. HDFS is used for storing huge data sets while Map Reduce is used for processing these huge datasets.
2. Hadoop consists of multiple concepts like COMBINER, PARTITIONER HBASE, PIG, HIVE, SQOOP to perform the easy and fast processing of huge data sets.
3. Hadoop is different from Relational databases and can process the high volume, high velocity and high variety of data i.e. BigData to produce the result.

In a Hadoop cluster, data is distributed to all the nodes of the cluster present on which data is stored as shown. Hadoop runs applications using the Map Reduce algorithm, where the data is processed in parallel with others. In short, Hadoop is used to develop applications that could perform complete statistical analysis on huge amounts of data.

The two core components of Hadoop are:

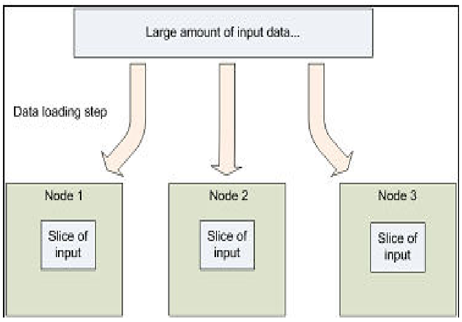

Fig.3. Hadoop cluster

A. *HDFS:*

The Hadoop Distributed File System (HDFS)[7] is based on the Google File System (GFS) and provides a distributed file system that is designed to run on commodity hardware. It has many similarities with existing distributed file systems. However, the differences from other distributed file systems are significant. It is highly fault tolerant and is designed to be deployed on low-cost hardware. It provides high throughput access to application data and is suitable for applications having large datasets. The Hadoop Distributed File System allows individual servers in a cluster to fail without aborting the computation process by ensuring data is replicated with redundancy across the cluster. There are no limits on the data that HDFS stores as it can be unstructured and schema-less.

B. *MapReduce:*

Map Reduce is a parallel programming model for writing distributed applications devised at Google for efficient processing of large amounts of data (multi-terabyte datasets), on large clusters (thousands of nodes)of commodity hardware in a reliable, fault-tolerant manner. The Map Reduce program runs on Hadoop which is an Apache open-sourceframework.

Hadoop minimizes the processing time as the files are distributed across different nodes in the cluster and these nodes work parallel thereby minimizing the processing time and increasing the performance. Programs must be written to a particular programming model, named "Map Reduce." In Map Reduce, records are processed in separately by isolated tasks called Mappers. The output from the Mappers is then moved together into a next set of tasks called Reducers, where results from different Mappers[8] can be merged together after shuffling and sorting as shown in fig. 4.

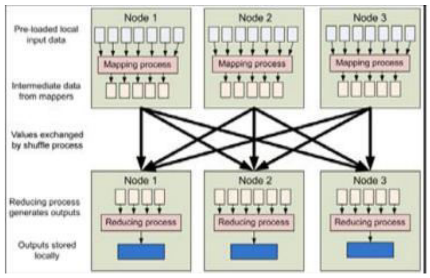

Fig.4. Mapping of records using Hadoop Distributed File System

V USE BIG DATA ANALYTICS TO DETECT APTS

Detecting APT within the firewall, and as they develop, is certainly not a simple task. In today's threat landscape, companies like J.P. Morgan must contend with extremely sophisticated intruders, who constantly change and refine their methods, as well as insiders who abuse legitimate access rights to manipulate and steal data.

In other words, both intruders and insiders may be hiding in relatively plain sight. The key, however, is to understand what is happening early on in an attack and on a continuing basis, evaluate the degree of risk at any one time, and have a plan to counter the activity.

Using big data analytics to detect attacks involves the following three interrelated and continuing steps:

1. Identify the risk:

Information-driven cyber intelligence allows entities to assess, manage, and minimize risks. Identifying and characterizing cyber threats and assessing the vulnerability of critical assets and operations specific to the threat helps organizations identify ways to reduce those risks and strategically prioritize risk-reduction measures. Unknown unknowns can be identified and managed, and the alarm can be raised early enough to act on unusual activity to counter attacks, preserve data, and protect customers and reputations. Organizations can plan for the likelihood and consequences of specific types of attacks and better manage and minimize the risk.

2. Use data analytics to detect threats and unusual behavior:

Data analytics can monitor patterns across a company's computer network, map what is normal activity, and detect previously unidentified APTs as manifested in anomalous occurrences in the network and devices. Analysts are alerted to suspicious connections between seemingly unrelated events or known entities of interest, as well as recurring visits from suspicious IP addresses or malicious domains. Again, IT and information security personnel are able to manage threats more effectively if they are detected quickly.

3. Prepare and activate an incident response plan:

When data analytics sounds the alarm early on, organizations and analysts must be ready to act quickly to prevent the compromise or loss of critical information. An incident response plan guides the organization in identifying APT's[11], recognizing risks and areas of vulnerability, and responding quickly to suspicious activity. This helps the organization take action to close the gap between attack and defense, stop the danger from spreading, and protect its operations and IP.

VI. CONCLUSION

In this paper, we propose a novel Approach for APT detection using Big Data Analytics with Hadoop as Service Anomaly by analyzing all datasets continually, filtering the data, recognizing patterns in Interactions to trigger an APT. Our security approach is good at detecting APT malware infections and is feasible for improving the sustainability of the system. The system processes advantages of high efficiency and accuracy. We believe that using Hadoop as service anomaly is a useful tool that can help to fight against cyber-crime especially theft of information from infected host.

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SUPERVISED LEARNING TO FILTER SHORT TEXT MESSAGES FROM ONLINE SOCIAL NETWORK USER WALLS

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Abstract: **Due to the rapid growth of world wide web, online social networking sites obtained overwhelming escalation and popularity around world. Customized Privacy policies and security measures made social network more attractive for an individual user. These features empower the users to control their private and public information, but user cannot avoid undesired messages written on their private or public walls. We realized how privacy is important to a user, so to avoid this Misappropriation in online social networking sites .We proposes a supervised learning approach which restricts unwanted content on user's walls. We make use of sentiment analysis to determine polarity of text. By using machine learning naïve bayes algorithm we classify text. This supervised learning approach automatically filters out messages from online social network user walls.**

Keywords—Sentiment analysis; Information retrieving; Information filtering; Machine learning; feature selection; feature extraction; Data preprocessing; naive bayes classifier.

I. INTRODUCTION

Now-a-days Social networking sites provide different services for users to communicate globally. Online social network (OSN) provides an opportunity for user to connect with people and interact with friends and families in a single environment. The main goal of social networking sites such as Facebook, Google, and twitter is to allow users to interact with each other online. Facebook has several features to attract an individual but some features have considerable impact on their private and public information. Facebook made drastic changes in which we share content with our friends, friends of friends and public. Users can state whose contents can be displayed on their walls by configuring custom settings. Facebook realized how privacy is essential to a user, so some changes were made to Facebook settings. Various restrictions are added to privacy setting but still users cannot control vulgar and offensive posts on their wall. Our main motivation is to restrict informal statements or messages in user walls. Messages are very short in length for which traditional text classification methods have limitations since short texts do not provide sufficient word occurrences .In order to prevent undesired posts on user walls, We propose a supervised learning approach to detect whether the posted message is good or bad. We adopt sentiment analysis to determine polarity of short text. Due to growth of social media sites sentiment analysis is very essential for analyzing customer behavior. Here we classify words into positive and negative class or neutral class. By making use of machine learning algorithm such as naive bayes. We calculate overall positive and negative score for the text if the documents contain more positive than negative score, it is assumed as positive document otherwise it is negative .In rare cases score is in between positive and negative, and then we consider it as neutral class. Our main goal is to determining whether a message is positive or negative. In section II we discuss about machine learning, in section III we see sentiment analysis .In IV we discuss how data processing is done and in section V we see how text is classified. In section VI we analyze how knowledge discover is done and finally in Section VII contains conclusion.

II. MACHINE LEARNING

According to Arthur Samuel, machine learning is defined as "field of study that gives ability to computer to learn without being explicitly programmed". Pattern recognition and computational learning theory are the applications of machine learning in field of artificial intelligence. A machine learning algorithm collects as many as possible examples and looks through these examples of previous work to predict. Machine Learning includes a number of statistical methods for handling classification and regression tasks with multiple independent and dependent variables. These methods include naïve bayes, k-nearest neighbor, support vector machine, decision tree etc. Machine learning is broadly classified into three different categories:

1) *Supervised learning*
2) *Unsupervised learning*
3) *Reinforcement learning*

A. Supervised learning

Supervised approach includes labeled classes from training dataset. It consist set of label classes input and their output are given and our goal is to learn a mapping from inputs to outputs.

B. Unsupervised learning

In this learning we give only inputs without any labels and our objective is to discover patterns in the data.

C. Reinforcement learning (RL)

Reinforcement learning is concerned with how software agents should take actions in an environment so as to maximize some notion of cumulative rewards.

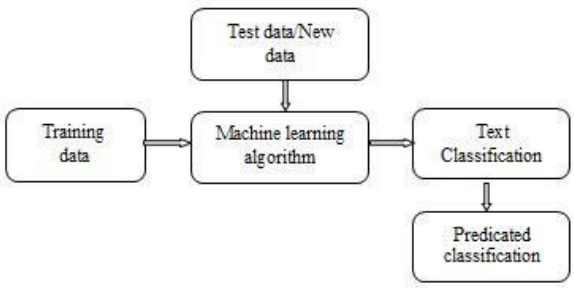

Fig. 1. Machine learning model

RL is one approach that can be taken for this learning process. An RL agent learns by interacting with its environment and observing the results of this interaction. These reduce primary way in which humans study performance. RL allows machine and software agents to automatically determine the ideal behavior within specific context, in order to maximize its performance.

III. SENTIMENT ANALYSIS

The use of natural language processing with an intension of deriving "sentiment" or "subjective information from text". Basically sentiment analysis is some kind of text classification. Extracting subjective information from a set of documents often uses reviews from online to determine polarity of specific object .Polarity defines the nature of the sentence, it can be positive, negative or neutral .It is useful for identifying public opinion in the social media. Sentiment analysis purpose is to determine the opinion and attitude of users towards some specific entity.

Sentiment analysis contains three essential elements such as source user, target user and polarity. Source user defines who expressed the text, target user specifies to whom sentiment is expressed and polarity gives nature of text. Sentiment analysis uses one of the machines learning approach to classify text as positive, negative or neutral. Text classification gained lots of importance in the field of information retrieval system. Mainly used to extract, identify, and characterize the sentiment of a text unit. Sometimes sentiment analysis is also referred as opinion mining. People express their opinions in complex way which can be easily comprehended by human but not by the machine. Since traditional techniques for classifying short text does not perform well, because term occurrence is very less. In this approach we use sentiment analysis to classify text that identifies polarity of the sentence. As we use bag of words model to classify word as unigram. Unigram words have no dependence between any two words. It is one of the general approaches for inserting words into a model. We use training data sets for classifying test data. Training data set must need to represent functional usage in the real world. We should choose what kind of data is to use for training set.

IV. DATA PREPROCESSING

Online social networking sites contain different data such as images, audio, video, shared locations, tags and textual messages. Heterogeneous data include noisy, incorrect and inconsistent data .Quality of data is also very essential. Low quality of data will lead low performance. Data preprocessing improve learning accuracy.

Preprocessing is very essential for classifying data because data may have futile words which do not provide any sense. Irrelevant and redundant data produce more difficulty in classification. Moreover data preprocessing reduces the complexity of text and makes easier to handle. After obtaining data we preprocess it before handling it to the classifier. Steps involved are as follows;

A. *Tokenization*

Document contains string of text which is not in perfectly ordered. Bag-of-words model represent text in a structured manner .This technique splits the text into sequence of individual tokens. For every blank space sentence is split into an individual word. The splitting of tokens is defined using all non-alphabetic characters. This resulted in tokens consisting of unigram .Unigram model outperforms all the other models. For example: "I really liked that movie" can be tokenized as [I, really, liked, that, movie].

B. *Filtering tokens*

Length based filtrating is applied for reducing the generated token set. To filter out the tokens having minimum length and maximum length. Therefore tokens with less than 3 characters and more than 25 characters will be discarded. Source and target users are identified and ignored. Hash tags (#), attherate (@), symbols, digits, punctuations, URLs are ignored. Stop words are also removed such as a, the, It, they etc., which do not make any sense in the statement. Here we have spell checking dictionary for checking incorrect words in text. Sentences also consist of a phrase which also specifies some sentiment about text. Classifying words into individual token we may also separate phrases. To avoid this we use a list of phrases along with their polarity. Phrases don't specify complete meaning of a word.

C. *Emoticons dictionary*

We prepare emoticon dictionary by assigning labels to each of the emoticons and classify it as either positive or negative. We use 170 emoticons that are listed on Wikipedia with their emotional state. In this day emoticons are mostly used for expressing their sentiment and opinion. For example ":=(" is labeled as negative whereas ":)" is labeled as positive. Replace all the emoticons with their acronyms by looking up the emoticon dictionary.

D. *Acronym dictionary*

An acronym dictionary has 6000 translated abbreviations. We replace short forms with their respective abbreviations by looking acronym dictionary .Some acronym can have more than one abbreviation .In such case, the acronym is translated according to context of sentence. For example, lol is translated to laughing out loud,gr8, gr8t as great, bff as best friend forever.

E. *Normalization*

For normalizing text we use a simple rule based model. For example we normalize gooood as good. Here we convert all words into lowercase for simplification. Data cleaning is also done to detect and correct irrelevant words. Negation plays a key role in expressing an opinion. Suppose if we have negation in the text then we prefix a negation before every word until the first punctuation or to the end of sentence. Reverse hypothesis is applied for negation to get actual meaning of word.e.g."He is not bad person" represent person is good. Actually "bad person" represent negative statement by applying reverse hypothesis "not bad person" changes to positive.

F. *Stemming*

Stemming defines a technique to find root (or) stem of a word. Porter's Algorithm is a process of removing suffixes from words in English .This algorithm is mostly used for stemming in Linguistics technology .That is getting rid of prefixes and suffixes .The filtered token set undergoes stemming to reduce the length of words until a minimum length is reached. This result in reduction of forms of a word into a single term. For example like, liking, liked as Lik, watching, watched as watch

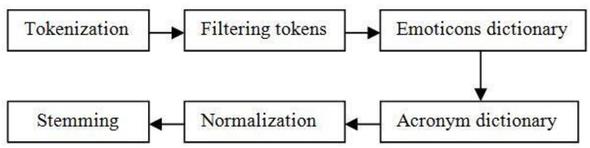

Fig. 2. Data processing

Table I. FREQUENCY TABLE

Terms	positive	negative
Complex	40	29
Problem	42	63
Solution	242	396

IV. TEXT CLASSIFICATION

Text classification of large document performs well but fails for small document. The bag-of-words model is mostly used model for text classification. This model represents text to be classified as collection of individual words with no dependence. Regardless of order of words and grammar content is classified. Unigram is a collection of individual words in the text to be classified .Each entry refers to frequency count of the corresponding word in the text. Using training data we label each term as positive or negative. In case if we don’t find any word in training data set then we look up the wordnet for synonyms and replace them. Wordnet is a lexical database for English language with an emphasis on synonym. We can get data resources from different social Media. The main purpose of sentiment analysis is to classify short paragraphic text. We count term frequency of each word in two classes (training model). Term based classification focus on classifying based on terms of the sentence .We classify the text into positive, negative or neutral. We use probability model naïve bayes text classifier. This classifier combines with decision rule generally known as “maximum a posterior”. Naive Bayes classifiers assume value of a particular feature is independent of value of any other feature. The naive bayes classifier gives an assumption that the independent variables are statistically independent. For text classification we need to select specific feature for training data. This can be done using any feature selection algorithms. This algorithm seeks for relevant features and discards irrelevant features. Feature selection is done for simplifying a model and reducing training time. Feature selection is different from feature extraction. Feature selection consist subset of features whereas feature extraction creates new feature from original features. Feature extraction is the first step in data preprocessing. Main advantage of naïve bayes classifier is, it requires very less training data set and scalable.

For calculating positive and negative of sentence,

$$C_k = \underset{k\in\{1,\dots,k\}}{argmax} P(c_k)\prod_{i=1}^{n} P\left(\frac{x_i}{C_k}\right) \tag{1}$$

Where $P(x_i/c_k)$ is probability of number of terms (x_i) in a text belong to a particular class (c_k) from i=1 to n. whereas $P(c_k)$ represents probability of class (c_k).C_k represents a class either positive or negative. Consider document which contain text as “most problems do have a solution”. After completing data preprocessing remaining terms of text are tabulated in term frequency table.

After preprocessing we get three terms i.e., complex, problem and solution. By training data set we have number of times a particular term has occurred in positive and negative statements which is known as term frequency. Naive bayes is primarily formulated for performing classification tasks. For every term in the sentence we calculate probability of the term in a particular class. It can be positive or negative class and calculate probability of positive and negative class. By considering the whole probability, we calculate likelihood of each term. Multiply the likelihood of terms with class probability which results in class C_k .Construct a likelihood table as below:

Table II. LIKELIHOOD TABLE

Terms	**Positive negative**		**P(x/pos) P(x/neg)**
Complex Problem Solution	40 42 242	29 63 396	0.1234 0.0594 0.1296 0.1290 0.7469 0.8114
Total	324	488	
	324/812 488/812 0.3390 0.6009		

For positive class

P(x/c)=p(complex/positive)=40/324=0.1234

P(problem/positive)=42/324=0.1296

P(solution/positive)=242/324=0.7469

P(positive)=324/812=0.3990

For negative class

P(x/c)=P(complex/negative)=29/488=0.0594

P(problem/negative)=63/488=0.1290

P(solution/solution)=396/488=0.8114

P(negative)=0.6009

Likelihood of positive class=0.1234*0.1296*0.7469*0.3990=4.7660x10^{-3}

Likelihood of negative class=0.0594*0.1290*0.8114*0.6009=3.7360x10^{-3}

Probability (positive) =0.5605

Probability (negative) =0.4394

After calculating likelihood we find the probability of positive and negative classes .Hence the probability of positive class is more than negative class therefore the statement is positive statement. The problem with naïve bayes classifier is that if term frequency of any word is zero then the entire solution will become zero. To avoid this we use Laplace estimator which add one to each count that we need would only make a negligible difference. These make considerable changes in the frequency table. Every word count value in the frequency table is incremented by one. In addition to positive and negative we can have one more class called neutral. Neutral class comes into picture when positive and negative classes have equal probabilities.

V. KNOWLEDGE DISCOVERY

In knowledge discovery classified text data is analyzed and their term occurrence is incremented in training data set. If text is positive then every word of that text is added to positive class in training data. This analyzed data help us in calculating new test data. Data changes from time to time so we need update training data set for better performance.

VI. CONCLUSION

In this paper we presented a supervised learning approach to filter short text messages from online social network user walls. We used naïve bayes machine learning algorithm for classifying text into positive, negative or neutral. The advantage of this classifier is that it requires less training data for classifying text and moreover it is scalable. By adopting this approach we can reduce informal text in online social networking user walls. This supervised learning filters undesired messages and gives total control of messages in their walls. This approach gives better privacy for users in online social networking site.

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HANDWRITTEN CHARACTER RECOGNITION USING MULTI LAYER FEED FORWARD NEURAL NETWORK

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Abstract: **The main task behind the making of this application is for Digitalization of character/signature system. This would be radically reducing the paper work on alphabetical characters and numerical. Characters will need to be transferred to locations where similarity matching needs to be done. This process to be done in paper work while transferring would mean a lot of time space. To avoid such a constraint and increase the efficiency of the recognition process this product is being progressed. Moreover the existing system of physically scanning the character or words in further instance is no where secure as human eye cannot predict accurately the similarity/differences in two images. Therefore such a situation needs to be avoided with the development of a more flexible algorithm using state of art technology and implemented directly. Faster recognition should also be reflected in the above process and overcome the time for recognition else scanning would be better option. This Software is used to identify and match the handwritten character of a user from previous knowledge of how his/her handwriting has been recorded. The handwriting firstly recorded will have characteristics of the same like curve, lineage etc identified using neural networks concepts of Multilayer Feed forward Network, the next on entering the writing can be identified and used for any specific purpose.**

Keywords- Digitalization, . Faster recognition, neural networks, Multilayer Feed forward Network.

1. PROBLEM DEFINITION

The main task behind the making of this application is for Digitalization of character/signature system. This would be radically reducing the paper work on alphabetical characters and numerical. Characters will need to be transferred to locations where similarity matching needs to be done. This process to be done in paper work while transferring would mean a lot of time space. To avoid such a constraint and increase the efficiency of the recognition process this product is being progressed. Moreover the existing system of physically scanning the character or words in further instance is no where secure as human eye cannot predict accurately the similarity/differences in two images. Therefore such a situation needs to be avoided with the development of a more flexible algorithm using state of art technology and implemented directly. Faster recognition should also be reflected in the above process and overcome the time for recognition else scanning would be better option.

2. SCOPE

The scope of this project is more than what it looks to be on final implementation. It can be used by any organization thriving for digital security for faster recognition of characters thus avoiding paper work or any other form of physical writing. Also Digital signature which currently is handled using manual scanning process and corrections can be made more precise using this application. This is will be due to the introduction of neural networks into the process. Also this project has the scope of being interfaced with devices to improve real time security using embedded systems. This software is not only used for security related services in verification code but on interfacing with related H/W devices the project can be developed to cater extensive security aspects in any field

3. EXISTING SYSTEM

A first approach has been the search for the criterions allowing the "absolute" identification of the writing elements, i.e. which would hold true for any handwritings whatsoever. This approach is successful only if the person who is writing would accept to comply with a very strict presentation, with well shaped letters of regular size and constant orientation. The present methods therefore do not allow the real time recognition of practically any handwriting whatsoever by making use of devices of reasonable powers such for instance as microcomputers.

4. PROPOSED SYSTEM

This project deals with developing an artificial neuron and creating a neural network using Microsoft Visual C#, make it learn using some algorithm to identify Hand written characters. This application consists of a '**Controls**' component which provides 'Drawing Box' and 'Letter Box' controls, and also defines a set of classes which can be used to develop applications which require handwritten (mouse-drawn) images as inputs. The '**OCR**' component makes use of created ANN and the 'Controls' component to recognize mouse-drawn capital letters of English alphabet.

5. HCR MODULES

The following modules are involved in the Handwritten Character Recognition System

- Neuron Creation Phase
- ANN controls component Unit
- Online Character Recognition

5.1 Neuron Creation Phase

This module involves creating an artificial neuron and developing a multi layered neural network. This process includes mainly the following functions

- **Declaration of Variables:** These include declaration of weights, threshold and other independent parameters for the Neuron Layers.
- **Declaration of Functions:** These include specification of Gaussian function, threshold function etc.
- **Random Generation of Weights:** These include generating random weights for each neuron layer. Mainly the classes included will be Neuron class for all the neuron specifications, Layer class handling all the array values and the neural network class.

Algorithm:

Step 1) Creation of Neuron class

```
Partial Class neuron
{
weights.thresholdvalue, min_range, max_range,w[i];
Activationfunction ()
{  GaussianFunction ();  }
randomgenerationweights()
{
w[i]=1.0…;
} } } }
```

Step 2) Creation of layer class

```
Class layer
{  neuron[];  }
```

step 3) Creation of Neural network Class

```
class{  layer[];}
```

5.2 ANN controls component Unit

This module involves developing an algorithm to scan, filter and make ANN learn about the hand written code. This is the module that is responsible for pattern searching and matching functionality. Once the neurons are created the they are considered to be blank spaces which need some information on it. For this the neurons are to be trained to certain specific characteristics from which they will be identified.

The main processes involved in ANN controls component is to scan and filter the neural network after training to recognize what kinds of input is being specified. The training of the neurons is done based on the input being specified in these case characters. Each time a neuron is to trained a character is represented In a format (supervised learning) and the neuron weights are adapted to those conditions. The values from each neuron layer are fed forward to the next layer using the feed forward neural network layer. The error correction is based on the back propagation algorithm which modifies the weight of neurons from the output activation function values so that the next time on every input the error percentage is reduced.

Initialize the weights in the network (often randomly)

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Do

For each example e in the training set

O = neural-net-output (network, e); forward pass

T = teacher output for e

Calculate error (T - O) at the output units

Compute delta_wi for all weights from hidden layer to output layer; backward pass

Compute delta_wi for all weights from input layer to hidden layer; backward pass continued

Update the weights in the network

Until all examples classified correctly or stopping criterion satisfied

Return the network

5.3 Online Character Recognition

This module involves creating a Windows application with graphical user interface components to configure neural network, to carryout learning process and to enable it to detect the hand written characters. These is the major functionality involved in the recognition process because this module involves screens where the user will be interacting for inputting the characters using the tools provided using Microsoft Visual Studio 2005. Some of these are briefly described below.

Drawing box: The drawing box is mainly provided to randomly script user characters to identify and train the neurons in the back end layers. The drawing box is a tool just like paint on which the mouse can be placed and any image can be drawn. It can retrieve various attributes such as pixel values(x, y coordinates) and strokes of mouse i.e. Mouse up or Mouse down.

Picture Buffer: The picture buffer is a frame for holding all the drawing components together. It allows the users to place their images on the drawing box within the specific boundaries.

6. HCR SYSTEM IMPLEMENTATION

Implementation is the stage where the theoretical design is turned into a working system. The most crucial stage is achieving a new successful system and giving confidence to the users by making the system work efficiently and effectively. The system can be implemented only after thorough testing is done and if it is found to work according to the specification.

It involves careful planning, investigation of the current system and its constraints on implementation, design of methods to achieve the change over and an evaluation of change over methods a part from planning. Two major tasks of preparing the implementation are education and training of the users and testing of the system. The more complex the system being implemented, the more involved will be the systems, analysis and design effort required just for implementation. The implementation phase comprises of several activities. The required hardware/ software acquisition is carried out. The system may require some software to be developed. For this, programs are written and tested. The user then changes over to his new fully tested system and the old system is discontinued.

The screen as in Fig1 showing what are the components involved on line Character Recognition process where in the user will be interacting for performing the Specific operations.

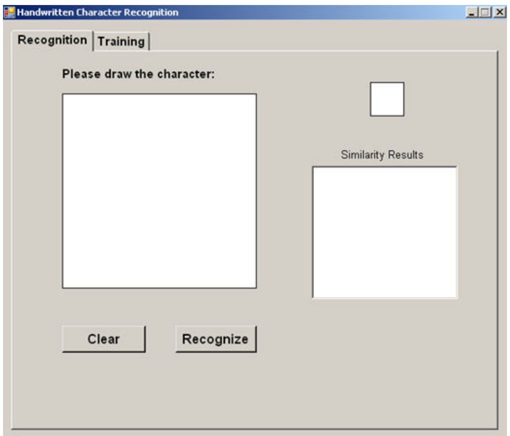

Fig1: The Basic outlook of the Project

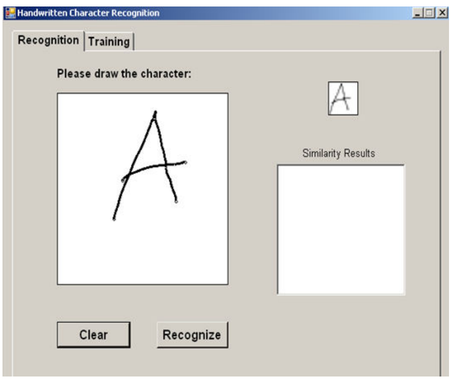

Fig 2: A character Inputted in the recognition Tab Screen.

The recognition screen as in Fig2 has a drawing screen as seen above on which the character will be marked as in this case 'A'. It provides us options to recognize the character and view similar results in the picture box seen on the right hand side of the screen.

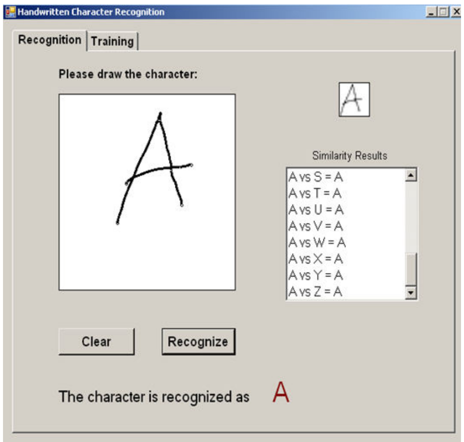

Fig 3: Recognized characters seen as Output.

The character recognition process can be seen in the Fig3. On matching with the trained set of characters the Output to the nearest character in this case 'A' is displayed.

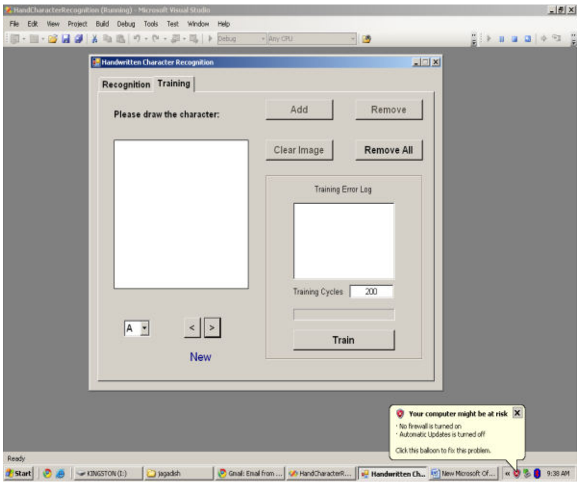

Fig 4: The training Tab screen

The initial process of training the neurons is shown in the fig4 with specific options including the adding a Character, removing a specific character and Removing all the trained characters. A character drawn will have similarity matches shown in the drop down box below the screen.

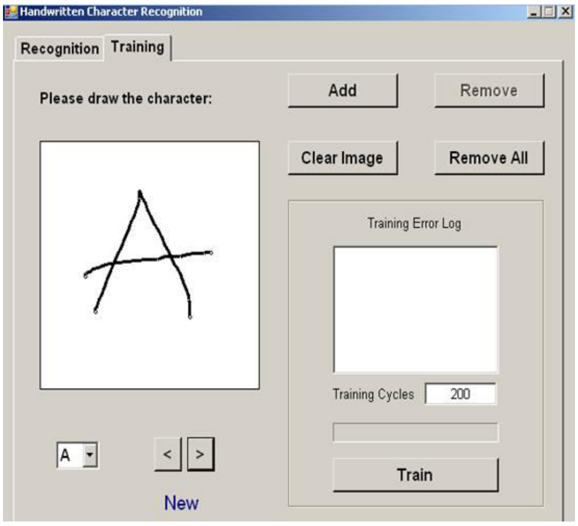

Fig 5: A rough image sketched for Training.

A character drawn will have similarity matches shown in the fig5 drop down box below the screen.

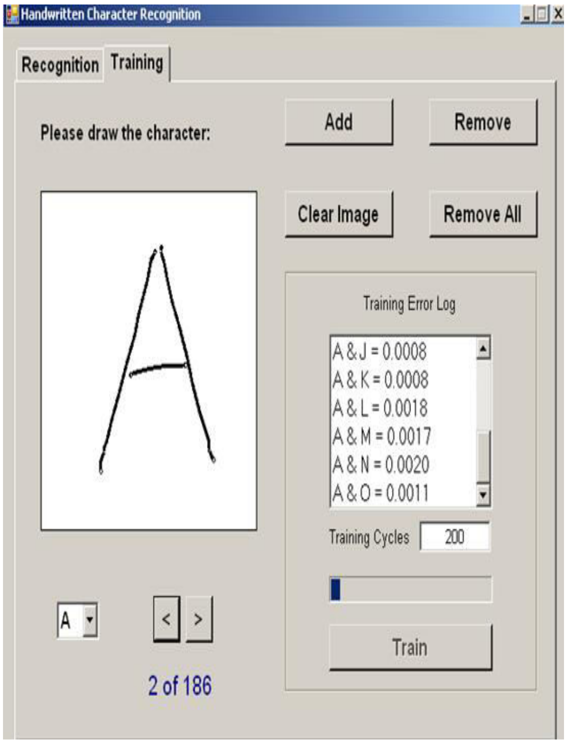

Fig 6: The training process being executed

The below drop down list showing the different character formats of A previously inputted .This one entered will added to one of them as shown in Fig6.

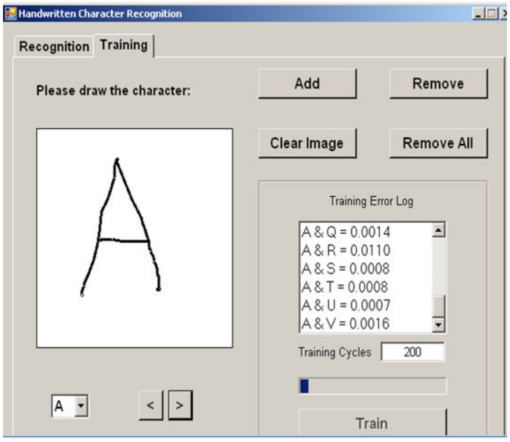

Fig 7: The retraining processing

In fig7 similar set of actions performed for repetitive training.

The screen as shown in fig8 shows the error percentage between any two comparisons.

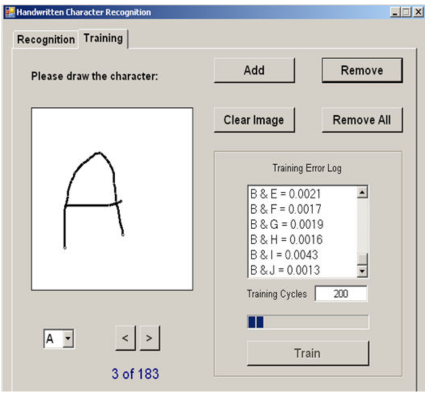

Fig 8: The Error log screen

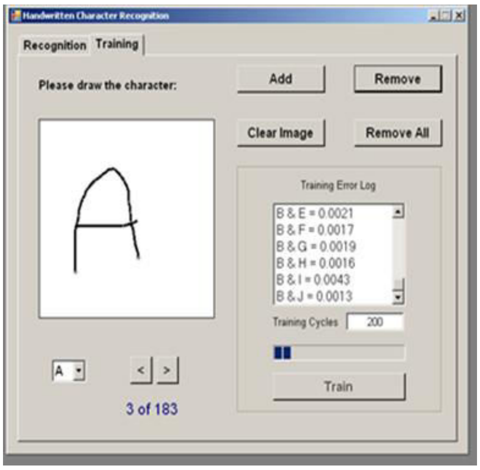

Fig 9: The online recognition process.

The online recognition process as shown in fig9 can actually be carried using devices which can scan out a character a Text into a screen. This could be Mouse, Scanner or OCR. In this case we have used a Mouse on the Windows .NET application.

7. TEST CASE

Test Case1: The training process of the characters is quite time taking and needs to be reduced to finish the entire cycle within restricted time to increase efficiency**.**

Test Case 2: The controls component process must be specifically tested to recognize characters which have not been added.

8. CONCLUSION

Recognition process is initiated with neuron creation. The created neuron is educated with inputs stored in Neurons (variables). Activation function responds with output values which will be handled by output function for further modification which will be the proceeding module. Once this is done the process of this module continues until the training of created neurons synaptic weights reaches a limit. Moreover the project has been designed to facilitate further improvements including extending to a real time scenario including Hardware devices specially Embedded Systems to improve the usability of it. Since the project has never been seem to have implemented in situations wherein such recognition process has been identified, it has been considered to be a Real time Project. **Future scope:** The future scope of this project will be to extend the current application beyond recognizing the individual letters to words & signature through pattern recognition.

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A STUDY ON MULTICAST ROUTING PROTOCOLS FOR MOBILE AD HOC NETWORKS

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***Abstract*:-The mobile Ad-hoc network (MANETs) defined as a collection of wireless nodes that are able to communicate each other without need of established infrastructure. MANETs follows multi hop routing from source to destination node or nodes. These networks have quite as many constraints because of uncertainty of radio interface and its limitations. The nodes in Ad hoc networks are battery operated with limited energy resources.**

A routing protocol uses routing algorithm for optimal network data transfer and communication between network nodes. Routing protocol facilitates router communication and overall network topology understanding. Each node in the network acts as a router and host and makes decision in forwarding packets from one node to another node.

Sending the multiple copies of packet to different nodes is called multicasting. Wired and infrastructure based wireless networks are supported by multicasting protocols. But applying this Concept in mobile Ad-hoc networks (MANETs) is a big challenge. Problems in Ad-hoc networks are scarcity of bandwidth, short life time of the nodes due to power constraints and dynamic topology due to mobility of nodes. These problems put forth to design a simple, scalable, robust and energy efficiency routing protocol for multicast environment. In this work multicast routing algorithms for MANETs are surveyed and categorized on the basis of metrics used for multicasting. These algorithms are analyzed in highlighting their strengths and deficiencies.

Keywords— MANETs, Multicast, Routing, Nodes.

1. INTRODUCTION

An Ad hoc network is a dynamically reconfigurable wireless network with no fixed infrastructure. Because of lack of infrastructure, centralized administration is not possible in ad hoc networks. Mobile ad hoc networks are self-organizing and self-configuring multi-hop wireless networks

Where, the structure of network changes dynamically.

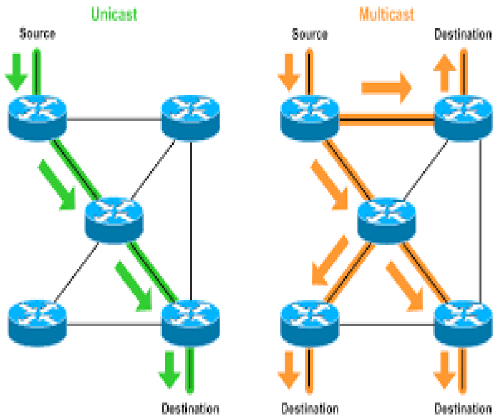

Fig. 1.1 Example of unicast and multicast d hoc networks

Communication among mobile nodes in an ad hoc network requires routing over multi-hop wireless paths. In MANETs all the mobile nodes play the role of a router. Even if the source and the destination mobile nodes are not within the communication range of each other, data packets are forwarded to the destination mobile node by relaying the transmission through intermediary mobile nodes.

Each device in a MANET is free to move independently in any direction, and will therefore changes its links to other direction of the devices frequently. Because of the limited transmission range of the nodes, multi-hops may be needed to reach each other nodes. Nodes in the networks utilize the same random access wireless channel and cooperating in a friendly manner to engaging themselves in multi-hop forwarding.

The primary responsibility of a routing protocol is exchanging the routing information by finding a feasible path to destination based on criteria such as minimum power required ,hop length and the life time of wireless link based on information about path breaks, processing power and bandwidth.

The primary purpose of multicasting is to carry out certain tasks that require point-to- multipoint, multipoint-to-multipoint voice and Data communication to a group of nodes.

Non-restricted easy deployment and mobility characteristics of MANETs make them very popular and highly suitable for various applications like emergencies, natural disasters, collaborative and distributed computing, military operations, civilian environments, commercial, wireless sensor networks, wireless Mesh networks, and hybrid wireless networks.

There are various protocols designed to minimize the design issues in mobile ad hoc wireless networks. All design areas can have some features and requirements for protocols in common. Uniqueness in the characteristics of an ad hoc network has several requirements for the routing protocol design. Ad-hoc routing must be simple, robust and minimize control message exchanges.

Multicast routing protocols which are used in static networks such as distance vector multicast routing protocol (DVMRP) do not perform well in ad hoc networks.

Generally in wireless networks multicast routing protocols are classified into two categories: tree based multicast routing protocols and mesh-based multicast routing protocols.

In tree based multicast routing protocols, there exist a single path between any sender-receiver pair. In Mesh based multicast routing protocols there may exist more than one path between a source- receiver pair. Mesh based protocols provide redundant (more than one) routes for maintaining connectivity to group members. However, tree-based protocols are not robust against frequent topology changes and the

Protocols can be assumed to be operating at unicast, multicast, geocast, broadcast situations.

In unicast routing protocols, one source transmits messages or data packets to destination.

Multicasting is a communication process in which the transmission of message (packets) is initiated by a single user and the message is received by one or more end users of the network.

Broadcast is a unique case of multicast, where all the nodes in the network should get broadcast message.

The purposes of geo cast protocols are to deliver data packets for a group of nodes which are situated on at specified geographical area.

Unicast is a special form of multicast and some multicast routing protocols supports both unicast and multicast routing.

2 Literature Survey

A mobile ad hoc network is a (MANET) is a collection of autonomous mobile nodes that forming a temporary network without aid of any stand alone infrastructure or centralized administration. Ad hoc wireless network uses shared radio channel and distributed routing fashion for communication in the network.

To send the data to a group of nodes in the ad-hoc network using unicast operation is very difficult. To avoid the difficulty Multicast operation is required. There are several multicast protocols which will play a key role in data transfer among the group of nodes.

packet delivery ratio (defined as the ratio of number of data packets delivered to all the receivers to the number of data packets received by all the receivers) drops at high mobility.

Ad-hoc wireless networks multicast protocols are classified into two types based on the type of operation: source initiated protocols and receiver initiated protocols.

In source initiate routing protocols the packet transmission is initiated by the sender. This is a 2 pass protocol (join REQ and join REP) for establishing the tree (mesh).

In receiver initiated multicasting protocols, the receiver uses flooding operation to search for paths to the sources of multicast groups to which it belongs. The tree or mesh construction is a 3 phase process (join REQ, join REP, join ACK).

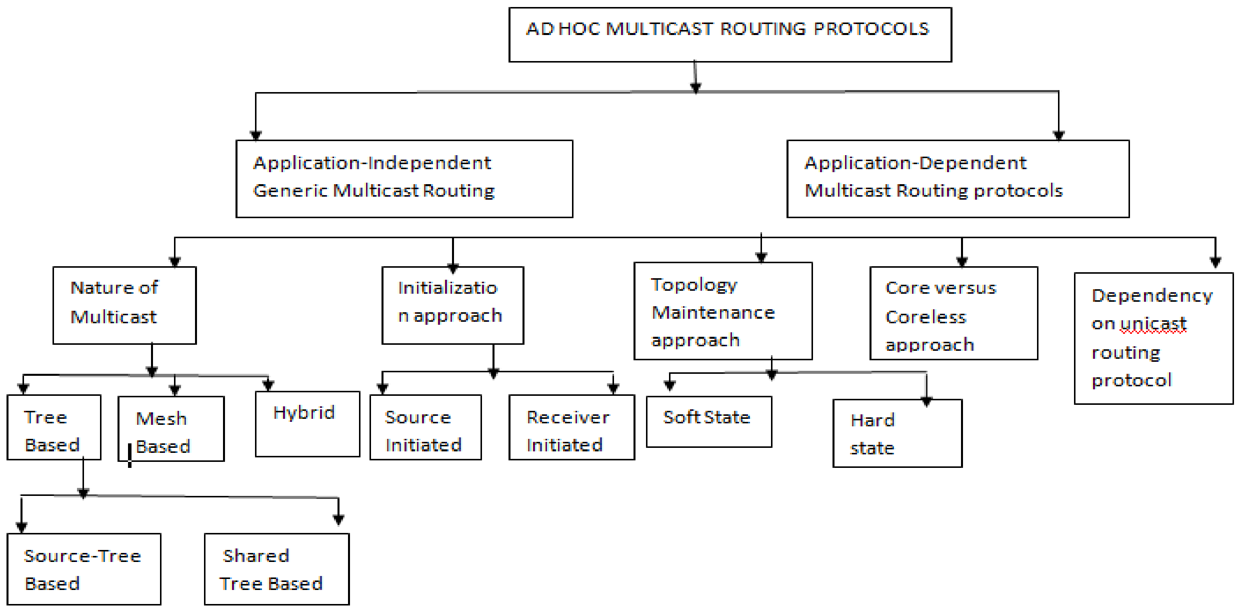

Fig 2.1: Classification of Multicast Routing Protocols over MANETS

2.1 Multicast topology classification

Ad hoc wireless networks multicast routing protocols can be broadly classified into two types:

Application-independent/Generic-multicast protocols and Application dependent multicast protocols.

Application-independent multicast protocols are used for conventional multicasting and application-dependent multicast protocols are meant only for specific applications for which they are designed.

Application-independent multicast protocols can be classified along different dimensions.

1. Based on topology
2. Based on initialization of the multicast session
3. Based on the topology maintenance mechanism
4. Based on the core versus core less approach
5. Base on the dependency on unicast routing protocols

Based On Topology:

Topology is defined as how nodes are arranged to form a network.

Based on the multicast topology current approaches used for ad-hoc multicast routing protocols can be classified into two types: tree-based and mesh-based.

Only a single path exists between a source-receiver pair in tree-based multicast routing protocols and more than one path may exists between a source-receiver pair in mesh-based multicast routing protocols.

Tree-based multicast protocols are more efficient compared to mesh-based multicast protocols, but mesh-based multicast protocols are robust due to the availability of multiple paths between the source and receiver. Tree-based multicast protocols can be further divided into two types: source- tree-based and shared-tree-based.

Generally, a sender initially performs flooding operation as a join message to all nodes in the network. The nodes which are interested can reply to the sender via the reverse path. After arriving all reply messages at the sender, a multicast tree rooted at the sender is formed. This type of tree construction is called sender-tree-based multicast protocols.

Usually a multicast group has several senders and thus which costs high for each sender to build its own tree. Some kind of protocols select a single sender to build a multicast tree that is shared with other senders. This type of tree construction is called a shared-tree-based multicast protocols and the selected sender is called the core node (group leader).

Other senders first transmit data packets to the core node and the core node then relays the packets downward the shared tree to all receivers. The type of initialization approach for tree construction by one or more senders is called a sender- initiated scheme.

The receiver-initiated scheme requires the receivers to initiate the tree construction, and it is often used for the shared-tree structure. Based on Initialization of the Multicast Session

The multicast group formation can be initiated by either the source or the receivers.

If the group formation is initiated only by the source node in a multicast routing, then it is called a source-initiated multicast routing protocol, and when it is initiated by the receivers of the multicast group, then it is called a receiver-initiated multicast routing protocol.

The multicast protocols which do not distinguish between source and receiver for initialization of the multicast group are called as source-or-receiver-initiated multicast routing protocols.

Based on the Topology Maintenance Mechanism:

Due to mobility of node, the routing structure requires reconfiguration. If a broken link is repaired by periodic flood packets issued by a sender, this kind of protocol is called soft-state maintenance. Periodic flooding packets also helps new members join the group.

If a link failure is repaired by a node on the link, this kind of protocol is called hard-state maintenance. Since no periodic flooding packets are issued in hard- state protocols, new members usually join the group by using expanded ring searches (iteratively expands the flood range). A group member usually leaves the group by sending a message to inform its parent of its departure. In addition to link failures, mobility of node may cause partition of the routing structure. For successfully delivering data packets to all group members' partition must be merged.

Each sender builds its own tree because sender-tree-based protocols incur higher control overhead than shared-tree- based protocols. Shared-tree-based protocols have two main disadvantages: sub-optimal multicast paths and single point of failure of the group leader. The group leader may locate in a bad position which further decreases multicast efficiency and increases packet latency. The mesh structure is robust against topological changes, but multicast efficiency is reduced. A new member cannot join a group as soon as it wishes in soft state protocol and hence it may miss interested packets for a while.

Based on the Core Versus Core Less Approach:

Core-based approach can be classified into two different approaches. They are dynamic core and static core approaches. If the current core node is failed then the member nodes of a multicast session elect or search for another one to be a new core then it is called dynamic core approach.

Static core approach differs from the dynamic core concept. A group of nodes or just one node controls all network tasks in static core approach. Network will be dropped due to any kind of failure of these core nodes.

Generally, core based routing protocols are used to reduce control overhead messages and to make a best utilization of bandwidth. However, they have a risk of a single point of node failure. Coreless based protocols solve the last problem but large overheads resulting of periodic announcements.

Based on the dependency on unicast routing protocols:

Multicast routing protocol had the ability to work as a multicast or a unicast protocol.

Separating unicast and multicast approaches has many disadvantages. It increases separated, redundant control overhead packets and it causes consequently wastage of bandwidth, decrease in overall efficiency of all the systems. Also, a complex problem is established when a unicast session need to be converted into a multicast session at any time.

Above all of these cons, it is a challenge for multicast protocol that relay on unicast one can work in heterogeneous networks.

Tree-Based Multicast Routing Protocols

Tree-based multicasting is a concept used in several wired multicast protocols to achieve high multicast efficiency. There is only one path between a source-receiver pair in tree based routing protocols. This tree based protocols are not robust enough to operate in high mobility environments.

Tree-based multicast protocols can be categorized into two types: source-tree- based multicast routing protocols and shared-tree- based multicast routing protocols.

A single multicast tree is maintained per source in sourced tree based multicast routing, where as a single tree is shared by all the sources in the multicast group in a shared-tree- based multicast routing.

Shared-tree-based multicast protocols are more scalable than source-tree-based multicast protocols.

Advantages:

- With the unicast route information, the multicast tree can be constructed more quickly and efficiently.
- It may incur very low overhead for a node to join or rejoin the session.
- It achieves higher multicast efficiency
- The path optimization process eliminates redundant paths gradually that leads to higher efficiency with lower packet transfer delay.
- It incurs low control overhead at low mobility.
- It can adapt to the change of mobility

Disadvantages

- Ease of tree structure fragile because of unpredictable topology changes due to mobility of nodes.
- Tree reconstruction delay and traffic concentration.
- Flooding Group Hello messages even if no sender for the group exists.
- Joining and rejoining of a node may take long time and waste much bandwidth since each node tries potential parent nodes arbitrarily.
- The usage of periodic beacons consumes bandwidth.
- The failure of a shared link affects several receivers
- If there are many receivers of the same multicast group, it leads to congestion in the most stable routes, which leads to increase in delay and a reduction in the packet delivery ratio.

- The occasional flooding of Advantages:
- Reduces data delivery latency during shortest paths.
- Lowers control overhead at mobility.

Mesh Based Multicast Routing Protocols

In ad hoc wireless networks, wireless channel breaks due to the mobility of the nodes. In case of multicast routing protocols, the path between a source and receiver suffers very much due to link breaks, which consists of multiple wireless hops. The protocols which provide multiple paths between a source-receiver pair are classified as mesh-based multicast routing protocols. Multiple paths in the network adds to the robustness of the mesh-based protocols at the cost of multicast efficiency.

Generally mesh-based multicast routing protocols robust due to the penalty of multiple paths between different nodes. But many of these proposals suffer from excessive control overhead which will affect on scalability and utilization of limited bandwidth, while others that apply core- based approach try to collect both robustness and efficiency from mesh and tree multicast approaches.

- Maintains a balance between routing efficiency and path robustness.

Disadvantages:

- Suffers from excessive flooding if there are a large number of senders.
- The duplicate transmissions waste bandwidth at low mobility.
- The failure of a core node affects several passive senders.
- High storage overhead is incurred for each node due to several maintained data structures.
- The periodic message exchanges among cores are a high overhead.
- Each member in the network waits for a period to select the best path which leads to take long time on mesh maintenance.

Protocol	Routing Approach	Unicast routing protocol	Loop-free	Route acquisition latency	Control packet flooding	Periodic control message	QoS support	Multicast Control Overhead
MAODV	Flat	Unicast-based (AODV)	Yes	High	Yes	Yes	No	Low
AMRIS	Flat	Autonomous	Yes	High	Yes	Yes	No	Low
BEMRP	Flat	Autonomous	Yes	High	Yes	No	No	Low
MZRP	Hierarchy	Unicast-based (ZRP)	Yes	High	Yes	Yes	No	Low
ABAM	Flat	Autonomous	Yes	High	No	No	No	Low
DDM	Flat	Dependent	Yes	High	Yes	Yes	No	Low
WBM	Flat	Autonomous	Yes	Low	Yes	No	No	Low
ADMR	Flat	Autonomous	Yes	High	No	No	No	Low
MCEDAR	Hierarchy	Unicast-based (CEDAR)	Yes	Low	Yes	No	Yes	High
PLBM	Flat	Unicast-based (PLBR)	Yes	Low	No	Yes	No	High
ASTM	Hierarchy	Dependent	Yes	Low	Yes	Yes	No	High
AMRoute	Flat	Dependent	No	Low	Yes	Yes	No	High
ODMRP	Flat	Autonomous	Yes	High	Yes	Yes	No	Low
DCMP	Flat	Autonomous	Yes	High	Yes	Yes	No	Low
CAMP)	Flat	Dependent (any Proactive)	Yes	Low	No	Yes	No	High
NSMP	Flat	Autonomous	Yes	High	Yes	Yes	No	Low
CAMP	Flat	Dependent (anyproactive)	Yes	Low	No	Yes	No	High
SRMP	Flat	Unicast-based (DSR)	Yes	High	No	No	Yes	Low

Table 1: Comparison of routing protocols in unicast and multicast approaches

Hybrid Multicast Routing Protocols

The Hybrid based multicast routing combines both the Mesh-based and the Tree- based routing approaches in order to achieve both robustness and efficiency.

3 Motivations

During the routing time Ad-hoc Networks have to suffer from many challenges. Dynamically changing topology and lack of centralized infrastructure are the biggest challenges in the designing of an ad- hoc network.

The position of nodes in ad-hoc network continuously varies due to mobility. We can't say that which particular protocol will give best performance in each and every case of topology variations very frequently. So we have to select a protocol which dynamically adapts to the ever changing topology very easily.

Another challenge in MANET is limited bandwidth. If we compare the wired network to the wireless network, wireless network has less and more varying bandwidth, so bandwidth efficiency is also a major concern in ad-hoc network protocols.

Limited power supply is the biggest challenge of an ad-hoc network so if we want to increase network life time (time duration when the first node of the network runs out of energy) as well the node life time then we must have an efficient energy management protocol.

Rapid increase in Ad-hoc network technology, wide deployment for several applications and the challenges that are facing in MANETs while packet transmissions in the network are the motivations to improve the performance in every case.

3. OPEN ISSUES AND ROBLEMS

Error-prone shared broadcast channel, limited bandwidth resource availability, limited energy resources with mobility of nodes, the hidden terminal problem and limited security make the design of a multicast routing protocol for ad hoc networks a challenging.

The current open issues and problems in ad- hoc networks are as follows:

- Energy saving
- Limited range in the wireless transmission
- Broadcast nature in the wireless transmission medium
- Packet loss due to transmission errors
- route changes in the network due to mobility induced
- Mobility-induced packet loss
- Battery constraints
- Frequent potential network partitions
- Ease of snooping on wireless transmissions
- Limited power supply

Information Storage:

- To store as less information as possible in the hosts.
- High storage overhead is incurred for each node due to several maintained data structures
- Suffers from excessive flooding when there is a large number of senders in the network

Messages Exchanged:

- Because the networks are bandwidth constrained, less exchange of information or messages between the nodes. Duplicate transmissions waste bandwidth at low mobility

Active Adaptability:

- The nodes should adapt themselves to mobility, power considerations, environmental conditions, etc.

Local Effect of Link Breakages:

- Network partitions or rapid movement in the mobile node.
- Unpredictable topology changes due to mobility of nodes.

Delay:

- Tree reconstruction delay and traffic concentration.
- Takes long time on mesh maintenance since each member waits for a period to select the best path.

4 Conclusion and Future Work

In this work, broad range of multicast routing protocols designed for MANETs is reviewed and.

classified all multicast routing protocols into two categories: tree-based protocols and mesh-based protocols. For each protocol, there are the properties, operations, strengths and weaknesses.

In this work more focus is only on general multicast routing protocols for ad hoc networks. There are other multicast routing protocols that aim at providing reliability, QoS (Quality of Service) guarantees, security, and so on.

Further investigation of these protocols and survey make more complete in future work to design a routing protocol with high packet delivery ratio, effective throughput, minimum delay with minimization of energy consumption levels.

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EFFECTIVE APPLICATIONS OF FILE SHARING IN MOBILE AD HOC SYSTEM

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ABSTRACT: ***File replication meant for proficient file sharing applications in mobile ad hoc networks was studied in recent times and several protocols of file replication were proposed for mobile ad hoc networks. Several properties of mobile ad hoc systems for instance node mobility, restricted communication range as well as resource, have provided several problems in recognizing of file sharing system. In the efforts of mobile ad hoc networks, random waypoint representation is generally used for normal mobile ad hoc networks and community-based mobility representation is used for disconnected mobile ad hoc networks. As a result, we make use of two models to symbolize two types of mobile ad hoc networks. In our work we initiate a novel thought of resource for replication of files that considers node storage as well as meeting frequency. We study influence of resource allocation on average querying delay and obtain an optimal file replication rule that allots resources to every file that is based on popularity as well as size.***

Keywords: File replication, Mobile ad hoc networks, Community-based mobility, Random waypoint, File sharing, Resource allocation.

1. INTRODUCTION:

The rising of applications concerning mobile file sharing will encourage study on peer-to-peer file sharing above mobile ad hoc networks. We make a consideration of mobile ad hoc networks such as normal as well as disconnected forms which are known as delay tolerant networks [1]. The former contains comparatively intense node distribution whereas latter contain lightly distributed nodes that meet up each other. The former contains comparatively intense node distribution whereas latter contain lightly distributed nodes that meet up each other. Sharing of files have turned out to be tricky as nodes within mobile ad hoc networks move about freely and exchanges data only when they are inside the range of communication. File replication is an efficient method to improve file accessibility and decrease file querying interruption. Unfortunately, it is not practical as well as ineffective to permit each node to hold file replicas within system that consider restricted node resources. File querying delay is constantly a most important concern within file sharing system. In mobile ad hoc networks, random waypoint representation is generally used for normal mobile ad hoc networks and community-based mobility representation is used for disconnected mobile ad hoc networks thus, we make use of two models to symbolize two types of mobile ad hoc networks [2][3]. On these two mobility models, our analysis respond on two assumptions such as: possibility of meeting an assured node is similar for the entire nodes as in random waypoint model or else the entire nodes in home community as in community-based model; nodes will move separately in both models of network. In our work we commence a novel thought of resource for replication of files that considers node storage as well as meeting frequency. We make a study of influence of resource allocation on average querying delay and obtain an optimal file replication rule that allots resources to every file that is based on popularity as well as size.

2. METHODOLOGY:

Applications concerning sharing of files in mobile networks have received increased attention in the recent times. The effectiveness of file querying will suffer from various properties such as node mobility, restricted communication range as well as resource. An instinctive process to lessen this difficulty is to generate file replicas within the network. In protocols of file replication for mobile ad hoc networks, each of the individual nodes will replicate files it queries commonly or else nodes will generate one replica for every file they query normally. In the former, redundant replicas will produce in system, thus wasting of resources and in the latter case although redundant replicas are decreased by group that is cooperation based, neighbouring nodes might disconnect from each other because of node mobility that leads to huge query delay. There are some works that handle caching of content in disconnected mobile ad hoc networks for proficient retrieval of data. Regardless of efforts, present protocols of file replication will lack a rule to assign restricted resources to files for creation of replica to attain lowest average querying delay. They consider storage as resource in support of replicas, but ignore that frequency of node to meet other nodes moreover influence file accessibility. Present efforts made on file replication in mobile ad hoc networks contains two limitations such as lacking of a rule to assign restricted resources to files for creation of replica to attain lowest average querying delay; and considering of storage as resource in support of replicas, but ignore that frequency of node to meet other nodes have an important function in file accessibility. In reality, node that contains superior meeting frequency by others will offer higher file accessibility and this will become more apparent in lightly distributed mobile ad hoc networks, where nodes will meet up disruptively [4]. We commence a novel thought of resource for replication of files that considers node storage as well as meeting frequency. We make a study of influence of resource allocation on average querying delay and obtain an optimal file replication rule that allots resources to every file that is based on popularity as well as size.

3. AN OVERVIEW OF PROPOSED SYSTEM:

File replication method improve file accessibility and decrease file querying interruption, unfortunately it is not practical as well as ineffective to permit each node to hold file replicas within system that consider restricted node resources. The local file sharing model of peer to peer will offer advantages such as enabling of file sharing when no base stations are obtainable; with peer to peer structural design, blockage on overloaded servers within present client server bass system of file sharing are avoided; exploiting of wasted peer to peer opportunities between mobile nodes. Hence nodes will freely access as well as share files in distributed environment of mobile ad hoc networks that support attractive applications. Mobile ad hoc networks refer to normal as well as disconnected forms which are known as delay tolerant networks. In research studies of mobile ad hoc networks, random waypoint representation is generally used for normal mobile ad hoc networks and community-based mobility representation is used for disconnected mobile ad hoc networks. As a result, we make use of two models to symbolize two types of mobile ad hoc networks. Recent protocols of file replication will lack a rule to assign restricted resources to files for creation of replica to attain lowest average querying delay and consider storage as resource in support of replicas, but ignore that frequency of node to meet other nodes moreover influence file accessibility. We commence a novel thought of resource for replication of files that considers node storage as well as meeting frequency. We make a study of influence of resource allocation on average querying delay and obtain an optimal file replication rule that allots resources to every file that is based on popularity as well as size. As some mobile ad hoc networks replication procedures we make use of the random waypoint model to model mobility of node within normal mobile ad hoc networks. In nodes of random waypoint model frequently move to particular point at random speed that means each of the node contain comparable probability to meet up other nodes that contain different probabilities of gathering nodes in reality [5]. We permit each of the nodes to contain randomly attained speed, to a certain extent than constantly altering speed as in normal model of random waypoint. The mobility model of community-based was used in content distribution for disconnected mobile ad hoc networks to represent node mobility. In this representation, complete test area is divided as various sub-areas, indicated as caves which holds one community. A node will belong to one or else extra community. The routines as well as social associations of node have a tendency to choose the pattern of mobility. Based on these two mobility models, our analysis respond on two assumptions such as: possibility of meeting an assured node is similar for the entire nodes as in random waypoint model or else the entire nodes in home community as in community-based model; nodes will move separately in both models of network. These two assumptions might not hold in actual cases, that limit applicability of analysis results on the other hand, analysis results will offer directions on replication of file since two models can correspond to important features in actual scenarios and were extensively employed in research studies. In optimal Replication of file with random waypoint model, we assume that inter-meeting time between nodes will follow exponential distribution. Then, possibility of meeting node is independent with earlier encountered node. As a result, we describe meeting capability of node as average number of nodes it meets up in unit instant and make use of it to examine optimal file replication. We consider meeting ability as well as storage in measuring of a node resource. In the optimal File Replication by means of community-basis mobility representation we carry out analysis in community-basis mobility representation. We consider node ability of satisfying then, number of nodes within community will represent number of queries in support of specified file generated in community consequently, file holder contain low ability to assure queries from a minute community [6].

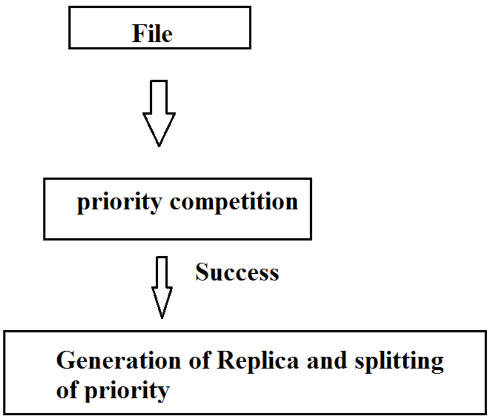

Fig1: Process of Replica distribution

4. CONCLUSION:

File replication intended for expert file sharing applications in mobile ad hoc networks was considered. Regardless of efforts on file replication, none of the research was performed on creation of global optimal replica by means of lowest average querying delay. Present works on file replication in mobile ad hoc networks contains two limitations such as lacking of a rule to assign restricted resources to files for creation of replica to attain lowest average querying delay; and considering of storage as resource in support of replicas, but ignore that frequency of node to meet other nodes have an important function in file accessibility. The approach of random waypoint representation is generally used for normal mobile ad hoc networks and community-based mobility representation is used for disconnected mobile ad hoc networks. As a result, we make use of two models to symbolize two types of mobile ad hoc networks. In our work we commence a novel thought of resource for replication of files that considers node storage as well as meeting frequency. We consider resource allocation on average querying delay and obtain an optimal file replication rule that allots resources to every file that is based on popularity as well as size. On bais of two mobility models, our analysis respond on two assumptions such as: possibility of meeting an assured node is similar for the entire nodes as in random waypoint model or else the entire nodes in home community as in community-based model; nodes will move separately in both models of network.

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TRUST AUTHENTICATION CLOUD SYSTEM BASED ON MULTIWAY DYNAMIC TRUST CHAIN MODEL

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Abstract: This paper sums up four security factors when analyzing co-residency threats caused by the special multitenant environment in the cloud. To secure the factors, a multi-way dynamic trust chain transfer model was planned on the premise of a measure interactive virtual machine and current behaviour to protect the integrity of the system. A trust chain construction module is designed in a virtual machine monitor. Through dynamic monitoring, it achieves the aim of transferring integrity between virtual machine. A cloud system with a trust authentication function is implemented on the basis of the model, and its practicability is shown.

Keywords: cloud computing; virtual machine; trustworthiness measurement; dynamic trust transfer

I. INTRODUCTION

With the development and wide application of cloud computing, cloud computing security [1-4] has progressively become a subject of concern. Cloud computing uses virtualization technology to achieve the encapsulation of underlying computing, network, and storage resources. A virtual machine is wide accustomed accommodate remote users and inevitably becomes the target of attacks. To provide reliable service to users, ensuring the trustworthiness and controllability of every virtual machine is important. Trusted computing [5] has introduced a replacement answer to cloud security. The appliance of trusted computing to cloud computing has become a crucial topic of analysis [6]. Santos 1st planned the conception of trusted cloud computing and designed a trusted cloud computing platform TCCP [7]. H.W Guo et al. [8] mentioned trusted computing model supported authorization. The essential security problems under the private model were solved by constructing a trust chain from the trusted root to any or all the virtual trusted parts.

In 2012, the team of Professor R.Y He planned a trust model referred to as TCTVM from the angle of cloud infrastructure security [9]. The main plan of analysis on these models is that the institution of static and dynamic trust chains to ensure system quality. The static trust chain was complete by extending the CRTM of TPM to a virtual machine monitor (VMM). The dynamic trust chain was complete on the premise of the static trust chain by extending VMM to vTPM. However, the trustworthy theme for the running method wasn't given once the user virtual machine started.

Having solely trusted virtual machine startup is insufficient [10]. A virtual machine doesn't exist in isolation. Cloud service suppliers often assign completely different virtual machines from different tenants to a similar physical machine in associate degree approach referred to as virtual machine co residency to with efficiency utilize physical resources. In 2009, Ristenpart et al. [11] of the University of California, in San Diego first pointed out the safety issues of virtual machine co-residency within the cloud. In recent years, a spread of co-residency security threats [12] are studied, together with resource interference, covert channel/side channel, denial of service, and load monitoring, that create the credibility of virtual machines troublesome to ensure.

To address the higher than issues, the establishing necessities of multiway trust chain are planned during this paper through the analysis of virtual machine co-residency security. This trust chain aims at a bidirectional management of input and output

I. DYNAMIC TRUST CHAIN IN CLOUD COMPUTING

1.1 Virtual machine co-residency security in cloud computing

Cloud computing platform could be a multi-tenant atmosphere. The virtual machine co-residency security issue is predicated on the belief that the users of cloud services (tenants) don't trust one another. Some malicious users attack common tenants and implement at-tacks on confidentiality on cloud platforms [13]. Kind of like common tenants, a malicious tenant will initiate and management multiple virtual machine instances. Given the chance that a malicious tenant is assigned to identical physical machine as common tenant instances, the malicious tenant will use shared physical resources (such as C.P.U., memory, disk, and network) to steal personal information or implement denial-of-service attacks. The co-residency threats to virtual machines square measure shown in Figure one. By dealings identical space and sort of virtual machine instances, malicious tenants will implement attacks on the victim's virtual machine and its platform. Currently, the threats to virtual machines square measure as follows: (1) Interference within the victim's virtual machine resources, as well as C.P.U., disk, and network resources; (2) Covert channel made by the shared resources among virtual machines; through such a channel, tip on the victim's virtual machine, like the RSA/AES key, is obtained; (3) Denial of service on the victim's virtual machine to scale back its convenience through the employment of a network transmission queue or vCPU computer hardware vulnerability; (4) Resource release. Through resource conflict between the co-resident virtual machines, the victim's virtual machine is forced to release resources; (5) observation and police investigation of the load condition of victims.

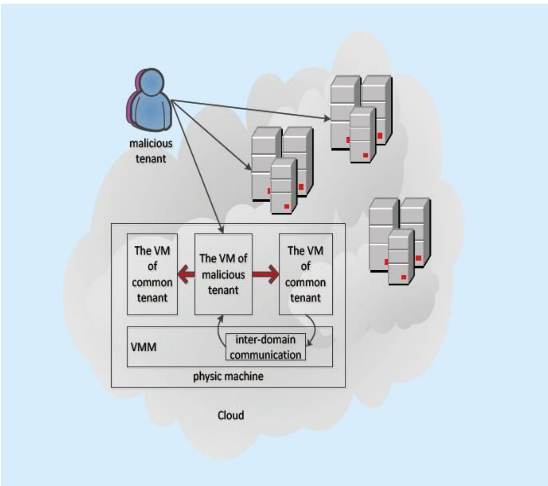

Fig.1 *Co-residency threats to virtual machine*

The on top of analysis of virtual machine security indicates that, within the cloud ADPS, the running security of a virtual ma-chine not solely depends on itself (usually the

Kernel and application components) however is additionally suffering from the surroundings (including VMM and different virtual machines within the system). The believability of cloud infrastructure services is set by the virtual machine and its interaction. The running security of virtual machines depends on four factors.

1. Integrity: The virtual machine has not been tampered with, and its own resources don't seem to be corrupted;
2. Interference: different virtual machines don't interfere with their standing.
3. Input: The input of the virtual machine is among the allowable varying. It protects the virtual machine from sudden security risks attributable to shared memory (covert channel);
4. Output: Output is distributed to a sure virtual machine to protect against malicious code intrusion and stop info from being used;

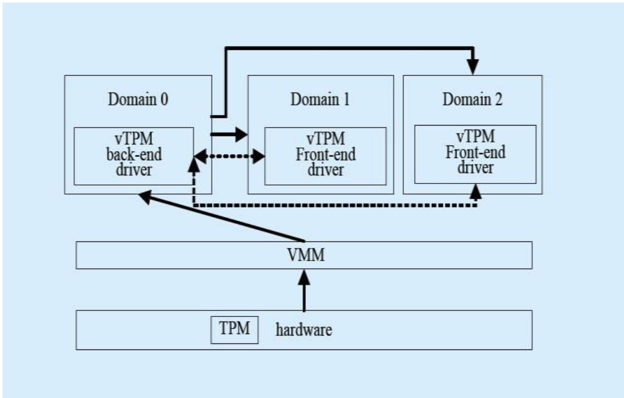

Fig.2 *Sketch map of longitudinal dynamic trust chain*

1.2 Analysis of dynamic trust chain transfer

The multi-tenant cloud surroundings places the virtual machine in danger, and different mechanisms [14, 15] are required to confirm period of time security. Threats to data are often self-addressed by determination the matter of knowledge system security. Thus, virtual machine dynamic trust chain transfer must be established.

The traditional answer of virtual machine metric within the cloud surroundings is to increase the only chain. The solid arrow in Figure two indicates a flow of sending trust. TPM and virtual technology are a ccustomed style a user-oriented vTPM (managed by the management machine Domain 0). The virtual security chip and therefore the virtual dependability root of every instance are measured by VMM, and therefore the trust chain is extended to every virtual machine and its application.

In the cloud computing surroundings, once the startup of the virtual machine, part loading, and different events occur less frequently, the system is in a very comparatively stable running state. The longitudinal trust transfer will then guarantee credibleness. However, as indicated within the previous section, this easy vertical trust chain cannot guarantee credibleness in real time whereas the virtual machine is running.

Once the virtual machine is started, the part is updated often, and it still desires a corresponding trust chain to confirm the protection of the virtual machine. Figure three shows the integrity transfer model planned

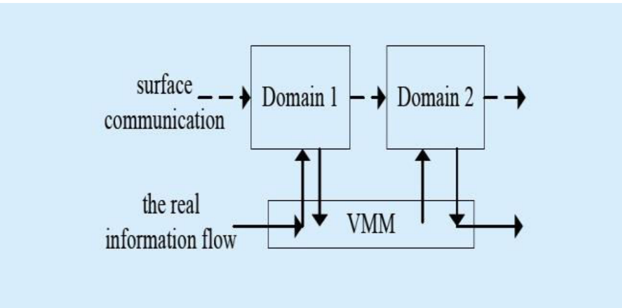

Fig.3 *Sketch map of horizontal dynamic trust transfer*

In Refs [10]. This model constructs a sure chain through lateral trust transfer, whose activity object includes these nodes solely. However, this model doesn't take consecutive module into consideration, thereby resulting in a range of security risks.
To surmount the disadvantages of existing trust chain transfer schemes, a multiway dynamic trust chain model (MDTCM) is pro-posed during this paper. This model combines horizontal and vertical trust transfer. The virtual machine is established credible through measurement on associate degree interactive virtual machine and also the current behavior. Integrity activity is conducted on the interactive virtual machine to realize longitudinal trust chain delivery. Moreover, it ensures data flow to the sure virtual machine. Current behavior measurement takes place between virtual machines and will reach horizontal trust transfer, thereby guaranteeing the integrity of knowledge supply and a reliable output.

As shown in Figure four, virtual machine Do-main1 performs associate degree action an on Domain2 to share dataD1. The trust chain construction (TCC) module detects the sensitive operation, and measures the integrity of Domain2 to confirm that it's invariably credible, thereby preventing the malicious use of knowledge. Additionally this behavior is measured. The model maintains the safety of the virtual machine through duplex management.

II. DESIGN OF MDTCM

2.1 Model description

To provide specific style ways and objectives of the model, seeable of the preconditions, this text presents the subsequent specific description:

1. TPM and vTPM technology provides a sure atmosphere for system startup. The sure base atmosphere is that the physical credible atmosphere, that isn't the con-tent of this text. The reader may talk over with the relevant literature.

2. Once the platform has believably started, the virtual machine still has the trust chain transfer, thereby making certain believability in real time.

3. Communication among virtual machines is often completed by MDTCM solely. No direct interactive communication takes place. The interaction between machine and also the underlying VMM is additionally completed by MDTCM.

4. MDTCM is clear to the upper-layer virtual machine.

2.2 Model structure

MDTCM adds the TCC module in the VMM. During trust transfer to the TCC component, the TCC component is responsible for dynamic monitoring. If sensitive behavior is triggered, then TCC executes measurement.

As shown in Figure 5, TCC includes the following three subcomponents:

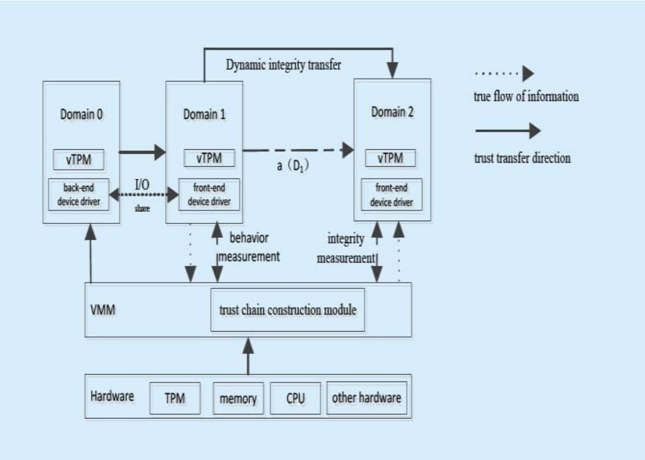

Fig.4 *Sketch map of MDTCM*

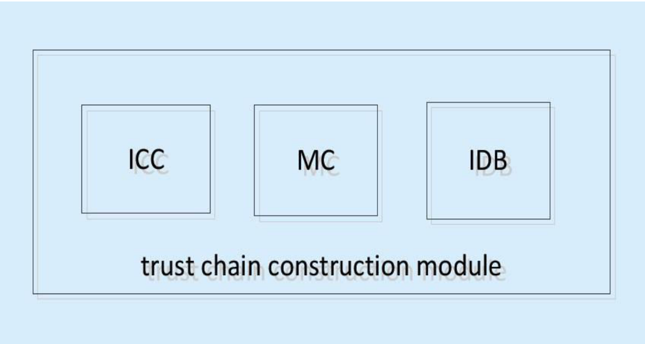

Fig.5 *TCC module*

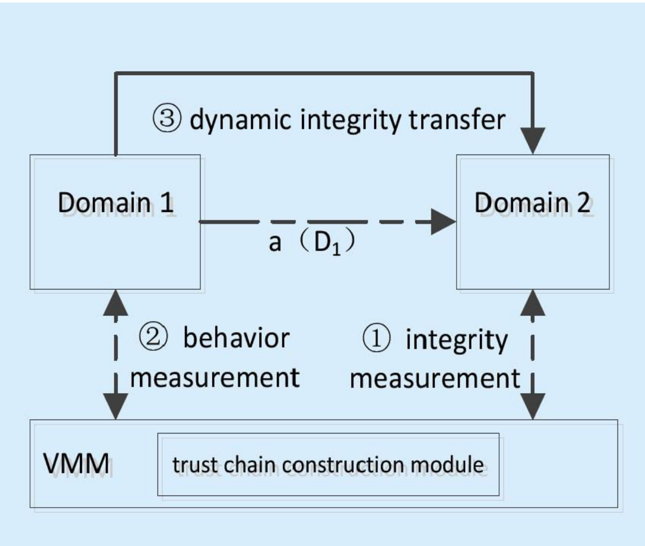

Fig.6 *Trust transfer mechanism*

1. Intercept and capture component (ICC) is the most active element in TCC. During the runtime of virtual machine, ICC monitors its behavior. Sensitive instructions are transferred to TCC using ICC.

2. Information database (IDB). When TCC starts, it opens up a certain space for IDB in privilege virtual machine Domain0, and IDB is empty. IDB only provides interface to MC. It can only be accessed and modified by MC. The component contains the following two types of storage information:

   a) In the installation of application, ICC will register its SHA-1 value in the IDB. ICC extends its vPCR.

   b) During virtual machine running, application are installed and loaded at any time. IDB maintains ALL (attribute label list) of each virtual machine. It keeps an account of the installed application and their order. The ALL of Domain1 can represented as:

3. Measurement component (MC). MC is the core processing unit of the TCC. After the judgment of ICC, the behavior that needs to continue to be dealt with will be measured by MC. MC has the function of query, comparison, and interaction with vTPM. Also it responsible for the fingerprint extraction of behavior to generate SBF (Sensitive Behavior Finger). It is responsible for all measurement tasks and judges the credibility of behavior through a verification process.

2.3 Trust transfer process

The trust transfer method takes totally into ac-count the protection factors, particularly the out-put. As shown in Figure VI, the transfer method is split into 3 steps:

1. Static integrity of the interactive VM Do-main2 is measured;

2. Believability of the present behavior is ensured;

3. Management is transferred.

The activity methodology fuses the static integrity activity and also the dynamic behavior activity. The static activity ensures the interactive virtual machine uninfected by the virus or replaced and provides a reliable initial state for the dynamic measurement. The new activity methodology compensates not just for the deficiencies for the static activity however additionally for the dynamic activity.

III. REALIZATION OF MDTCM

3.1 Integrity measurement

Integrity measurement focuses on each component of the interactive machine. The integrity measurement mechanism is shown in Figure 7. After the judgment of ICC, the sensitive operation is transferred to MC. MC analyzes the behavior to initiate the measurement process. In the installation of the user application, MC is also responsible for measuring the application code. All the SHA-1 values consist of the VM's static data, and recorded by IDB.

This measurement is carried out with the help of physical TPM. However, the conduct of dynamic integrity measurement piggybacks on the vTPM. When sensitive instruction is transferred, MC sends a metric instruction to the vTPM of an interactive virtual machine.

The vTPM conducts the measurement, thereby increasing efficiency and alleviates the strain on the physical TPM. The measurement results are compared with standard SHA-1 values. If they are equal, then the interactive VM is static credible. Otherwise it is incredible.

3.2 Behavior measurement

According to the definition given by TCG [16], whether the computing platform can be trusted directly depends on its behavior, which should be consistent with the expected strategy. The detailed realization and behavior measurement processes are shown in Figure 8.

First, the AMS (API function invoking Module Sets) of VM is constructed. It consists of AAM (Application API invoking Module set). $AMS_1 = AAM_{11} + AAM_{12} \ldots$ With the help of array ALL1, AMS1 is generated.

Second, from the point of view of system security, two types of operations are selected as monitoring points.

1. The operation to make a change to the boundary, such as the creation and destruction of the VM and the expansion or reduction of the VM boundary.
2. The cross-border instruction that is the access and modification operation initiated by the other VM. After real-time behavior is monitored, the SBF is matched with the expected behavior of the VM. The credibility of the real-time

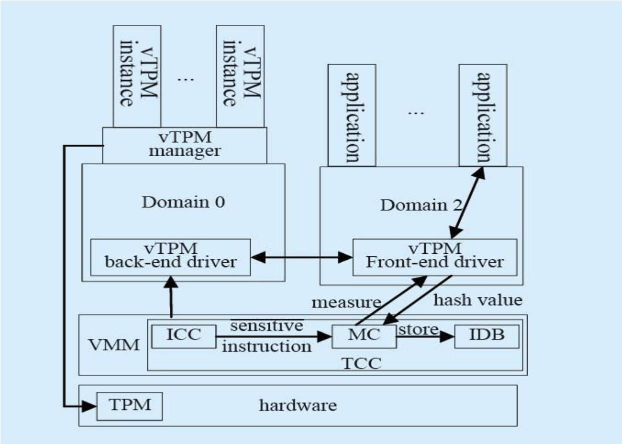

Fig.7 *Integrity measurement mechanism*

Behavior is authenticated through the matching algorithm. The verification process is discussed in detail in the next section.

3.3 Trust authentication cloud system based on MDTCM

On the basis of MDTCM, the cloud system is developed with dynamic trusted authentication. On account of behavior credibility, the dynamic trusted authentication function

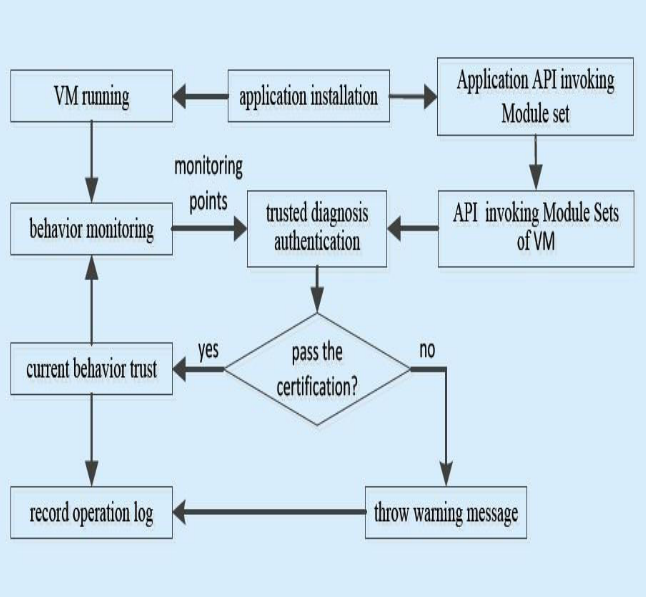

Fig.8 *Basic idea of behavior trust authentication*

Is added to maintain the dynamic credibility. The cloud system includes two parts, namely, cloud server and trust authentication party. As shown in Figure 9, the secure channel between the two parts is deployed on the basis of SSL. The cloud nodes are configured with open-source VMM system Xen, and inlaid with TPM chip. Dynamic monitoring is carried out by using the hook kernel function of SELinux security architecture. The ICC component is the specific implementation of the hook function interface.

Trust authentication party can be divided into four modules, namely, AM (Application Manager), VMFG (Virtual Machine Finger-print Generate), BFMD (behavior Fingerprint Match Decision) and HA (History Audit).

Specific verification process

MC→Domain2 measure

$P_{21}, P_{22}, P_{23}\ldots$

get

$H(P_{21}), H(P_{22}), H(P_{23})\ldots$

$H($ query ALL_2

calculate $I_2 = H(P_{21}) | H(P_{22}) | H(P_{23})\ldots$ query IDB

compare $I_2 = SV$? (Standard Value)

if true, send array ALL_1 and SBF

or, throw error message

Part 2: Trust Authentication

Party download ALL_1 and SBF

$\left.\begin{matrix} ALL \\ AM \end{matrix}\right\} AMML(AMM\ \text{List})$

$AMML \rightarrow AMS_1 = AAM_{11} + AAM_{12} + AAM_{13}\ldots$

$AMS_1 \xrightarrow{VMFG} VMF$

BFMD: int fl ag=1;

for (int i=0; i<={ l(VMF)-l(SBF)+1}; i++)

{

for (int j=0; j<=l(SBF); j++)

{ if (sbf[j] ⊕ vmf[i]==0) { fl ag=0; break; } else continue;

}

if (fl ag=0) break;

}

if (fl ag) return true;

else return false;

Trust authentication party executes policy decision and log audit to monitor the virtual machine running situation and block anomaly detection. If authentication is passed, trust authentication party return the information that behavior is credible. Then control is transferred, and the system keeps running credibly.

The above verification process indicates that the cloud system-based MDTCM can monitor the operation of virtual machines in real time. Moreover, it authenticates and

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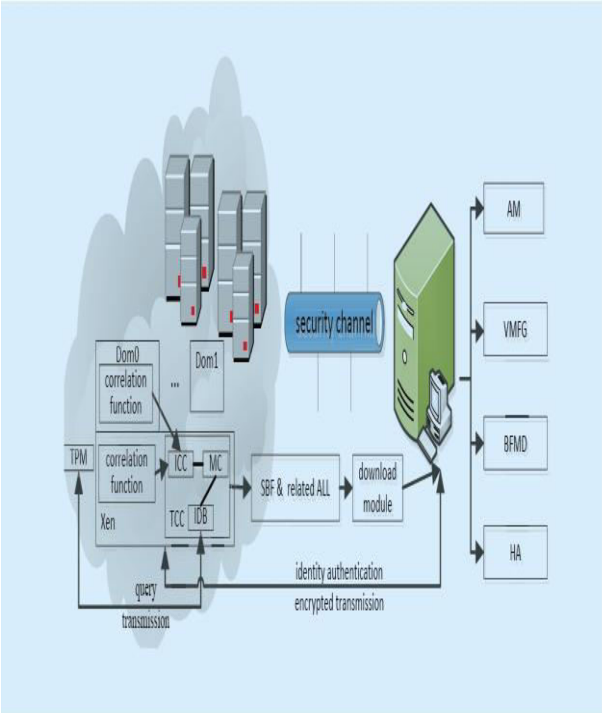

Fig.9 *Overall architecture of the cloud system*

Controls the operation of virtual machines dynamically based on behavior fingerprint and behavior authentication. Cloud system security and dynamic trust in a multi-tenant environment is guaranteed by securing the interactive virtual machine and the current behavior.

IV. CONCLUSIONS

The model guarantees the real-time credibility and communication security of a virtual machine. A cloud system-based MDTCM is built to show the feasibility of the virtual ma-chine. MDTCM can better reflect the dynamic characteristics of the virtual machine, there-by providing a good basis for studies on the security of virtual machines. Future research should consider strengthening the defense of information databases, which have a high confidentiality requirement and may be subjected to concentrated attacks.

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CONSISTENCY AS A SERVICE MODEL: A LARGE DATA CLOUD AND MULTIPLE SMALL AUDIT CLOUDS

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Abstract — Cloud storage services have become very popular due to their infinite advantages. To provide always-on access, a cloud service provider (CSP) maintains multiple copies for each piece of data on geographically distributed servers. A major disadvantage of using this technique in clouds is that it is very expensive to achieve strong consistency on a worldwide scale. In this system, a novel consistency as a service (CaaS) model is presented, which involves a large data cloud and many small audit clouds. In the CaaS model we are presented in our system, a data cloud is maintained by a CSP. A group of users that participate an audit cloud can verify whether the data cloud provides the promised level of consistency or not. The system proposes a two level auditing architecture, which need a loosely synchronize clock in the audit cloud. Then design algorithms to measure the severity of violations with two metrics: the commonality of violations, and the oldness value of read. Finally, heuristic auditing strategy (HAS) is devised to find out as many violations as possible. Many experiments were performed using a combination of simulations and a real cloud deployment to validate HAS.

Keywords- Cloud Service Provider (CSP), Consistency as a Service (CaaS), Heuristic Auditing strategy, Service Level Agreement, User Operation Table, Directed Acyclic Graph, Network Time Protocol

INTRODUCTION

Clouds computing has become more popular choice, because it has succeeded in giving guaranteed basic services like virtualized infrastructure system and providing data storage, etc. e.g. Amazon, SimpleDB are example of such systems. The customers or end users by making use of these services, become authorized users and able to access the data from anywhere and at any time using any device and getting confidence that the capital investment is going to less. The cloud service provider popularly known as CSP promising the users data is going to be available as 24/7, and they can access it efficiently. The CSP stores the different copies of data in a distributed fashion on different servers, which geographically present in different places. The main issue with distributing multiple copies of data called as replication technique is resultant into a very expensive process to provide strong consistency operation. In the coming days user is assured to see the latest updates about this service or operation. Many cloud service providers provide week consistency, we call such consistency as eventual consistency, where a user can read the data for particular time. Now-a-days stronger consistency assurance is getting importance. Consider the following figure. In the above figure data is stored in multiple copies on five cloud servers (CS1, CS2…, CS5), users specified in the figure share data through a cloud storage service. Here the cloud should provide casual consistency service, where a user Alice uploads a data on the cloud server CS4. Here the user update should be reflected in all the servers. If cloud service provider provides only eventual consistency then receiver user is going to receive the old version of data. Such a integrated design based on traditional version may not satisfy customer requirements.

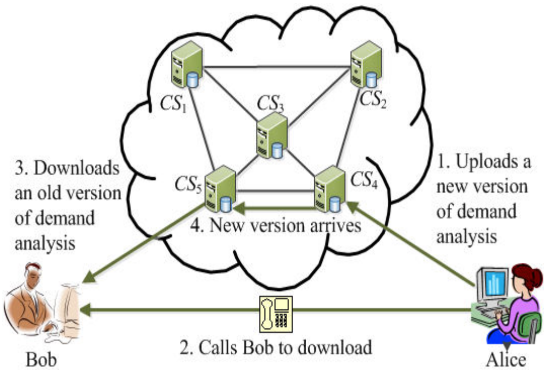

Fig. 1. An application that requires causal consistency

Hence we can conclude that different applications need different level of consistency operation. In this work, we propose CaaS model as a ideal consistency operation, which the applications of today's date are demanding. The standard CaaS model consists of large cloud data created by CSP and complete the operation scheme it contains many audit clouds which are formed by authorized group of users. These group of users working on a project and making a document, which constantly checking whether data cloud offers a guaranteed level of consistency or not. This standard model incorporating two-level auditing structure, which involves a synchronized clock assigning tasks to audit cloud and then performing global auditing with global trace of standard operations periodically by auditor chosen from an audit cloud. Local auditing is going to be performed and focuses on continuous read and read-your write process consistencies, which is going to be done online by a light-weight algorithm, while global auditing operation focuses on casual consistency, where it constructs a directed graph. Whatever graph has been constructed, if it directed acyclic graph, which is also called as precedence graph, we can say that casual consistency is maintained. We confirm the severity of violations by calculated two metrics for the standard CaaS model: One is called commonality of violations and other is staleness of value of read operation. Finally in this work, we propose a approach called Heuristic Auditing Strategy(HAS), which proves cloud consistency and required cost i.e. actual cost per transaction. The two level auditing structures basically contain 2 auditing types: 1. Local Auditing 2. Global Auditing 1. Local Auditing: structure each user can perform local auditing with local trace operation periodically .this auditing focuses on monotonic read and read your write consistency .which can be perform by light-weight online algorithm the local auditing algorithm is online algorithm. 2. Global Auditing: the auditor can be selected from audit cloud .the main works of the auditor is to perform global auditing with global trace operation .this auditing focuses on causal consistency because causal consistency perform by constructing directed graph .the directed acyclic graph is constructed then causal consistency is obtain .Finally we propose analytical auditing strategy which appropriate reads to reveal many unsuccessful result.

Although the infrastructures under the cloud are much more powerful and reliable than personal computing devices, they are still facing the broad range of both internal and external threats for data integrity.

Second, there do exist various motivations for CSP to behave unfaithfully toward the cloud users regarding their outsourced data status.

In particular, simply downloading all the data for its integrity verification is not a practical solution due to the expensiveness in I/O and transmission cost across the network. Besides, it is often insufficient to detect the data corruption only when accessing the data, as it does not give users correctness assurance for those un accessed data and might be too late to recover the data loss or damage.

Encryption does not completely solve the problem of protecting data privacy against third-party auditing but just reduces it to the complex key management domain. Unauthorized data leakage still remains possible due to the potential exposure of decryption keys.

LITERATURE SURVEY

S. Esteves [15] has highlighted the work on the critical data information, which is stored in cloud data centers across the globe, and getting increased in great way. And they are using different replication methods or approaches to deliver high-availability of services, demand of high performance, and mainly to maintain the consistency among multiple copies of data i.e. replicas. The proposed technique targets or focuses on data stored in tabular format, provides rationalization of resources, here bandwidth means bandwidth and which also requires improvement in the QOS parameters like latency value, performance in the network and availability of resources.

H. Wada [16] has introduced a new class of data storage systems, called NoSQL (Not Only SQL), has emerged to complement traditional database systems, with rejection of general ACID transactions as one common feature. Here the authors have brought a new area of for study, where a new class has been introduced in the data storage system, which is called as NoSQL (Not Only SQL), which have been introduced to support the traditional or general classic database systems, where it come across the removal of ACID transactions properties from these systems, considered as the one common general feature.

M. Rahman [18] highlighting the study of storage systems, these systems with large-scale key-value standard storage systems, compromise with consistency for the interest of dependability, i.e.

availability of resources and partition tolerance systems, as well as performance of network, considering latency parameter. The system under this study provides eventual consistency, which is difficult to implement in real time systems. The authors have attempted to measure such consistency empirically, but these systems suffer from some drawbacks. But their accuracy has been limited due to some state-of-the are systems considering consistency benchmarks.

D. Kossmann [19] discussed about now-a-days the cloud computing systems have many advantages for deploying applications in real time systems, example for such applications are data-intensive applications. Under this system, the user follows pay-as-you-go service model, under which user pay to the services, which he has used. The system provides a promising service with reduced cost. Another promising feature service to add here is, the system provides unlimited throughput by adding servers. They have focused on the transaction processing work, such as read and update workloads, instead of on the other processing like analytics operation or OLAP workloads operation.

E. Brewer [3] proposed the work on common current distributed systems, even the ones sytem that work, tend to be very generally fragile: they are very hard to keep up, very hard to manage, hard to grow, very hard to evolve, and very hard to program. Here in this talk, the author looks at several problems in an attempt to clear the way we think about these general systems. These problems include the general fault model, very high availability, considered graceful degradation, specific data consistency, specifying evolution, composition operation, and autonomy process.

L. Lamport [20] has introduced the concept of formalism for generally specifying and particular reasoning about concurrent systems has been described. It is not going to be based upon specific atomic actions. A general definition of a higher-level system is given and also justified correspondingly. And considering in Part II, the generic formalism is going to be used to specify several specific classes of inter-process communication and algorithms have been proven to be correct for implementing them.

SCOPE OF THE PROJECT

The scope of this project is to upload and download a file from cloud. While providing cloud consistency, the following objectives are to be met:

1] Understanding the novel consistency as a service (CaaS) model provided by the cloud service provider.

2] The cloud computing solution should provide basic consistency as service.

3] Maintain synchronized clock at audit clouds that responsible for checking weather cloud provide promised consistency or not.

4] Service Availability

EXISTING SYSTEM

- By using the cloud storage services, the customers can access data stored in a cloud anytime and anywhere using any device, without caring about a large amount of capital investment when deploying the underlying hardware infrastructures.
- The cloud service provider (CSP) stores data replicas on multiple geographically distributed servers.
- Where a user can read stale data for a period of time. The domain name system (DNS) is one of the most popular applications that implement eventual consistency. Updates to a name will not be visible immediately, but all clients are ensured to see them eventually.

DISADVANTAGES:

- The replication technique in clouds is that it is very expensive to achieve strong consistency.
- Hard to verify replica in the data cloud is the latest one or not.

PROPOSED SYSTEM

- In this paper, we presented a consistency as a service (CaaS) model and a two-level auditing structure to help users verify whether the cloud serviceprovider (CSP) is providing the promised consistency, and to quantify the severity of the violations, if any.
- With the CaaS model, the users can assess the quality of cloud services and choose a right CSP among various candidates, e.g, the least expensive one that still provides adequate consistency for the users' applications.

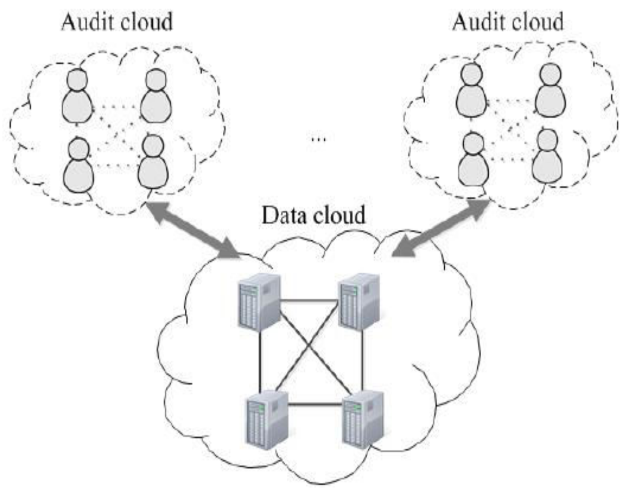

Fig. 2.Consistency as a service model.

ADVANTAGES :

- Do not require a global clock among all users for total ordering of operations.
- The users can assess the quality of cloud services.
- choose a right CSP
- Among various candidates, e.g, the least expensive one that still provides adequate consistency for the users' applications.
- As a rising subject, cloud consistency is playing an increasingly important role in the decision support activity of every walk of life.
- Get Efficient Item set result based on the caas.

IMPLEMENTATION

1. System Module:-

In the first module, we develop the System Module with User Module, Admin Module, and Auditor Module. In user module, the authorized user should undergo registration operation andregister their detailed information and collect the secret key information for login operation and user can able to upload the file regarding the operation like auditing. Next in user module, system user already uploaded files can be stored in system cloud database in systematic manner. Next Auditor can view the file or locate the file present in the database and it can be very secured. In admin module admin will be able to view all the register user details; and also user uploads information details, and third party TPA activities regarding the auditing strategy. In auditor module, the generally selected auditor can do the auditing operation based on the strategy operation called as heuristic auditing strategy. This operation is related to the basic operation of document verification. Then special unit Auditor can collect and check the auditing file, the he decide whether to reject or accept the file. On this he is going to make a report and enter all the details and about the decision like whether it's good or bad. Also in this module an auditor can submit revision report. In this report information like accept or waiting. After the decision if status is present as a accept means then the user can view the file else if the status is waiting condition means then user cannot view the particular file.

2.User Operation Table: In this module each user is going to maintain a UOT for the operation of recording local operations in systematic manner. Then each present record in the UOT unit is shown or explored by three components: first is an operation parameter, next the present logical vector, and finally more important one is physical vector. When the user is working on any operation, he is going to record his complete activity and also the current logical vector and final value as a physical vector, in his own UOT. In this module each user of the system is going to maintain a special logical vector and a basic physical vector to track the complete logical and physical time when an operation is going to take place correspondingly.

3. Local Consistency Auditing: Local consistency auditing technique is an online algorithm. The operation in this module or unit, in which each user is going to record all his complete activities and store in his UOT. During the read operation, the authorized user is going to perform local consistency operation in an independent manner.

4. Global Consistency Auditing: Global consistency auditing technique is going be considered as an offline algorithm. Next to consider is that an auditor periodically will be selected from the audit cloud system to perform the special operation like global consistency auditing technique. Hence in this case the auditor is going to collect all users' UOTs for obtaining a special global trace of all activities. Then later executing global auditing technique, selected auditor is going to send results of auditing operation as well as its vectors values to all other authorized users . Now given the auditor's vector values, then each user will come to know other users' new clocks up to next global auditing.

RELATED WORK

As per our survey there are many previously worked done in the field of cloud data consistency. In paper [1] they reduce that confusion by clarifying terms, providing simple figures to quantify comparisons between of cloud and conventional Computing, and identifying the top technical and non-technical obstacles and opportunities of Cloud Computing. We believe the only plausible solution to very high availability is multiple Cloud Service Providers (CSP).We predict Cloud Computing will grow, so developers should take it into account. Regardless whether a cloud provider sells services at a low level of abstraction like EC2 or a higher level like AppEngine, we believe that computing, storage and networking must all focus on horizontal scalability of virtualized resources rather than on single node performance. In paper [2] they provide the better way to store any file on cloud storage. A key contribution of COPS is its scalability, which can enforce causal dependencies between keys stored across an entire cluster, rather than a single server like previous systems. But storing a file on a cluster is producing huge problem for providing consistency. Measuring consistency is a very important task in this system because monitoring and controlling consistency is major goal of proposed system. By the time various benchmarking techniques are offered.

In [3] they are providing whole new perspective to see the need of consistency as a service (CaaS).In cloud computing storage services, every service request has an associated cost. In particular, it is possible to assign a very precise monetary cost to consistency protocols (i.e., the number of service calls needed to ensure the consistency level times the cost per call). Therefore, in cloud storage services, consistency not only influences the performance and availability of the systems but also the overall operational cost. In [4] to know the consistency models we have studied local and global consistency model of Dengyong Zhou, Olivier Bousquet, and Thomas NavinLal. The key to semi-supervised learning problems is the consistency assumption, which essentially requires a classifying function to be sufficiently smooth with respect to the intrinsic structure revealed by a huge amount of labeled and unlabeled points. We proposed a simple algorithm to obtain such a solution, which demonstrated effective use of unlabeled data in experiments including toy data, digit recognition and text categorization.

CONCLUSION

In this paper, The presented system is a consistency as a service (CaaS) model and a two-level auditing scheme to help users validate whether the cloud service provider (CSP) is providing the promised consistency, and to enumerate the occurrences of the violations. The CaaS model used in the system helps the users can assess the superiority of cloud services and decide a right CSP among various services. For example the less costly one that still provides satisfactory consistency for the users' applications.

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AN INNOVATIVE SECURITY PRINCIPLE TO PREVENT BITTER ATTACKS IN MANETS

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ABSTRACT: As ad-hoc network uses wireless communication link between the mobile nodes due to this it's less strong to attacks. The Sybil attack could be a mainly dangerous threat, where a person node illegitimately claims numerous particulars in mobile random systems. In Sybil attack, attackers use several particulars anytime or they take-off identity of some reliable node inside the network. Wireless systems are more inclined to many attacks. A specific attack could be a black hole attack by which malicious node wrongly proclaiming it acquiring the brand new and least path to the destination. In insufficient a priori reliable nodes the invention and confirmation of fellow citizen positions becomes mainly challenging in the feel of competitors pointing at harming the unit. We present and appraise the Secure Message Transmission protocol, which safeguards the information transmission against arbitrary malicious behavior of other nodes. Peak could be a lightweight, yet extremely effective, protocol that may operate exclusively in a finish-to-finish manner. It exploits the redundancy of multipath routing and changes its operation to stay effective and efficient during highly adverse conditions. Peak is more preferable suitable for support QoS legitimate-time communications within the random networking atmosphere. The safety of understanding transmission is accomplished without limited presumptions across the network nodes' trust and network membership, without requiring invasion recognition schemes, at the expense of moderate multi-path transmission overhead only.

Keywords: Mobile ad-hoc networks (MANET), lightweight solution, Sybil attack, secure message transmission (SMT) protocol.

I. INTRODUCTION

MANET is definitely an autonomous system includes numerous nodes. These nodes talk to one another through wireless links. Because of infrastructure less nature of MANET so that as there's no central authority to keep and control the network causes it to be susceptible to various attacks. There's a panic attack which in turn causes lot destruction to some network known as Sybil attack [1]. Inside a MANET, each node not just functions as a host but could also behave as a router. While receiving data, nodes likewise need cooperation with one another to forward the information packets, therefore developing a radio LAN. Essentially MANETs have two sorts i.e. open and closed systems. Closed MANETs don't provide any type of open access. Hence all nodes which are creating the network are pre-distributed through some type of identifications that are necessary toward joining the network. However open MANETs have no type of well-defined limitations i.e. anybody can join and left the network anytime. This issue in open MANET security i.e. to supply defense against selfish and malicious nodes within the network operation. Sybil attack is among the vital and challenging condition in open mobile Adhoc systems. Sybil attack is really a serious threat to those systems because of will need a distinctive, individual and chronic identity per node plus lack of central identity management. A Sybil attacker can impact towards the random systems in lots of ways. The communication in mobile random systems comprises two phases, the path discovery and also the data transmission. Within an adverse atmosphere, both phases are susceptible to a number of attacks. First, opponents can disrupt the path discovery by impersonating the destination, by responding with stale or corrupted routing information, or by disseminating forged control traffic. To supply comprehensive security, both phases of MANET communication should be safeguarded. It's significant that secure routing methods, which make sure the correctness from the discovered topology information, cannot on their own make sure the secure and undisrupted delivery of sent data. The routing process may disrupt because of the collaboration attacks by malicious node in MANET. The malicious node could cause security problems like grey hole and collaborative black hole attacks. Many research people say concerning the prevention and recognition of MANET. Because of the dynamic topology the mobile nodes will face several attacks. The commonest attacks are black hole and grey hole attacks. There are two kinds of attacks in MANET. They're active attack and passive attack. The growing development of wireless mobile network and Position verification system services requires in which the nodes can be found within the unstructured systems which means this process easily discover in which the actual nodes should be put into the mobile network system as well as discover adversarial nodes. To discover the reliable nodes, within this paper we have to talked about concerning the secure data transmission within the mobile network with verification of position by utilizing NPV formula and CRT formula. The NPV performs majorly three procedures in mobile network i) Safely figuring out own location, ii) Secure neighbor discovery, and iii) Neighbor position verification. It's mainly concentrate on to perform against a number of different colluding attacks. After NPV process the CRT formula determines the large data that may be divided by a few given divisors. The drawbacks in existing system are: Foe can fool the protocol simply by proclaiming false locations. This sensor could be assaulted by utilizing fake id nodes [2]. Another disadvantage to the presented option would be that every node only has a nearby view that may not be enough to reliably identify all position faking node. One method to counter security attacks is always to cryptographically safeguard and authenticate all control and knowledge traffic. But to achieve this, nodes would want the way to establish the required trust associations with every single peer they're transiently connected with, including nodes that simply forward their data. To retain the data transmission phase, we advise and assess the Secure Message Transmission (Peak) protocol, a finish-to-finish secure data forwarding protocol tailored towards the MANET communication needs. The Peak protocol safeguards pairwise communication across a mystery frequently altering network, possibly in the existence of opponents that could exhibit arbitrary behavior.

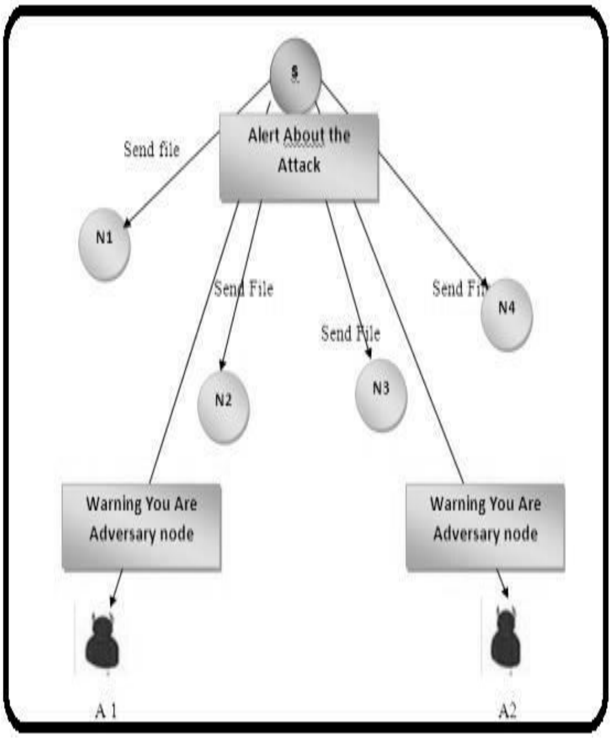

Fig.1. Block Diagram of Attacks

II. PREVIOUS STUDY

The authors S. Abbas, M. Merabti, D.L. Jones, and K. Kifayat described, the distinction between a new legitimate node and a new Sybil identity can be made based on their neighborhood joining behavior. For example, new legitimate nodes become neighbors as soon as they enter inside the radio range of other nodes; hence their *first* RSS at the receiver node will be low enough. In contrast a Sybil attacker, which is already a neighbor, will cause its new identity to appear abruptly in the neighborhood [3]. When the Sybil attacker creates new identity, the signal strength of that identity will be high enough to be distinguished from the newly joined neighbor. The authors J.M. Chang, P.C. Tsou, I. Woungang, H.C. Chao and C.F. Lai discussed, in mobile ad hoc networks, a primary requirement for the establishment of communication among nodes is that nodes should cooperate with each other. In the presence of malevolent nodes, this requirement may lead to serious security concerns; for instance, such nodes may disrupt the routing process. In this context, preventing or detecting malicious nodes launching gray hole or collaborative blackhole attacks is a challenge. This paper attempts to resolve this issue by designing a dynamic source routing DSR-based routing mechanism, which is referred to as the cooperative bait detection scheme that integrates the advantages of both proactive and reactive defense architectures. Our CBDS method implements a reverse tracing technique to help in achieving the stated goal. P. Papadimitratos and Z.J. Haas proposed, the vision of nomadic computing with its ubiquitous access has stimulated much interest in the Mobile Ad Hoc Networking technology. However, its proliferation strongly depends on the availability of security provisions, among other factors. In the open, collaborative MANET environment practically any node can maliciously or selfishly disrupt and deny communication of other nodes. In this paper, the authors present and evaluate the Secure Message Transmission protocol, which safeguards the data transmission against arbitrary malicious behavior of other nodes. SMT is a lightweight, yet very effective, protocol that can operate solely in an end-to-end manner [4]. It exploits the redundancy of multipath routing and adapts its operation to remain efficient and effective even in highly adverse environments. SMT is capable of delivering up to 250% more data messages than a protocol that does not secure the data transmission. Moreover, SMT outperforms an alternative single-path protocol, a secure data forwarding protocol they term Secure Single Path (SSP) protocol.

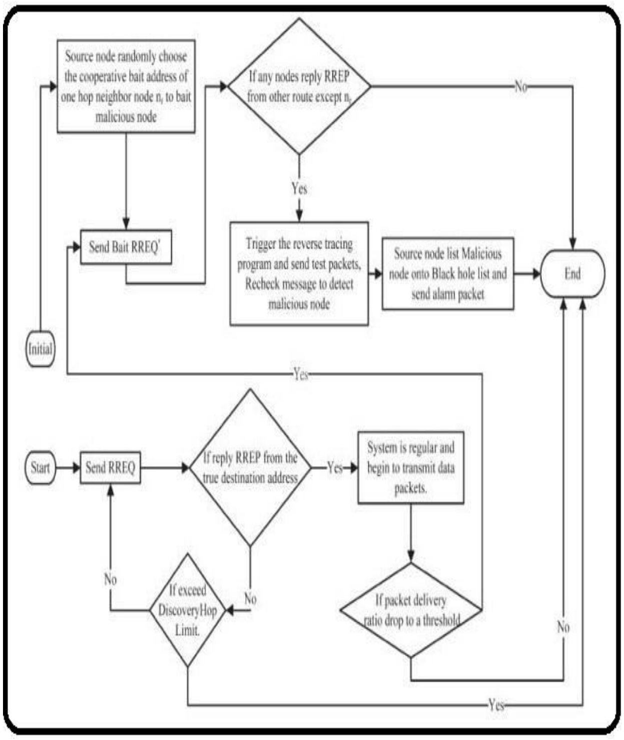

Fig.2. Data flow of CBDS

III. METHODOLOGY

We're finding the existence of malicious Sybil node within the route discovery phase. We used three orthogonal dimensions i.e. direct v/s indirect communication, fabricated v/s stolen particulars, and simultaneity for the recognition of Sybil nodes [5]. Essentially there are two strategies to validate a status. The main one strategy is direct validation in which a node directly tests once the identity of other node is relevant otherwise but another strategy is indirect validation where already verified nodes are permitted to ensure or refute other nodes. Within the suggested identity verification and resource based algorithmic approach the important thing factor pool defense can be used direct validation of nodes. Lightweight because it doesn't use almost every other hardware or antennae because of its implementation. It's acquainted with identify Sybil Attacks. It offers three steps: Kinds of Sybil nodes: Threshold value: Comparison. In above lightweight technique, sometimes there's Sybil nodes whose speed is under 10m/s which nodes are detected as legitimate nodes. To get rid of this issue with above technique, it's modified. In above lightweight technique just one parameter can be utilized i.e. speed. Ideas added two more parameter i.e. energy and frequency. With such two parameters it's giving better results than previous. The 2ACK plan enables you to reduce the hacker's effect in malicious node. The overall theme of 2ACK formula is, when the source node S transmits the packets for the neighbor hop effectively, the destination node D of neighbor decide to send a distinctive two-hop acknowledgement is known as 2ACK to specify the packets received effectively. The 2ACK transmission can be used only a couple of data packet transmission. During this paper fully spread cooperative request NPV, which enables a node, hereinafter known to as verifier, to uncover and validate the job from the communication fellow citizen is planned [6]. This paper handles a transportable random network, in which a ubiquitous infrastructure is absent; along with the position data needs to be got through node-to-node communication. This type of scenario is of certain interest because it leaves the doorway uncovered for adversarial nodes to ill use or interrupts the region-based services. The suggested approach is ideal for free adhoc environs, and, consequently, it doesn't rely on the presence of a reliable structure or in the priori reliable nodes and furthermore it controls cooperation but concurs a node to attain all verification techniques individually. During this suggested technique we enhance our work by talking about using china Remainder Theorem for nearly any novel forwarding plan in Mobile adhoc systems fond of mixing low computational complexity and satisfaction. The suggested approach is characterized getting a computationally simple packet splitting procedure able to reduce the energy required for transmission [7]. The suggested technique executes a splitting within the original messages in lots of packets to make sure that each node within the network will forward only small packets.

IV. CONCLUSION

Security is among the foremost issues in MANET. Within this paper an answer is suggested to identify the existence of the attacker within the Route discovery phase. In lightweight Sybil attack recognition technique just one parameter speed can be used to identify the malicious node but may there's Sybil nodes whose speed is under 10m/s will also be detected as legitimate nodes. To supply more security, strategy is suggested by which two more parameters are utilized the brand new technique cooperative bait recognition plan can be used within this paper to beat the grey hole and black hole attacks. The CBDS formula is merges to kinds of recognition mechanism namely positive and reactive. Within this paper, a superior approach continues to be suggested. Experiment demonstrated our protocol is extremely helpful for data transmission in mobile ad-hoc network and it is prevent colluding attackers. The outcomes make sure option would be active in discovering nodes advertising false position. The CRT formula determines the large data that may be divided by a few given divisors. And also the divided data will be transmitting towards the node through the various path after which finally collecting the divided data and merge that data and undergo the destination node. The Peak protocol would be to retain the data forwarding operation for MANET routing methods. Peak protocol uses topological and transmission redundancies and utilizes feedback, exchanged only backward and forward interacting finish-nodes. By doing this, Peak remains effective even under highly adverse conditions.

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DESIGNING A PRIMITIVE TO CONFIRM THE RELIABILITY AND COMPETENCE OF OUTSOURCED DATA

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ABSTRACT: Although scientific study has suggested many dynamic PoS schemes in single user environments, the issue in multi-user environments is not investigated sufficiently. Dynamic Evidence of Storage (PoS) is really a helpful cryptographic primitive that allows a person to determine the integrity of outsourced files and also to efficiently update the files inside a cloud server. To the very best of our understanding, no existing dynamic PoSs supports this method. An operating multi-user cloud storage system needs the secure client-side mix-user Deduplication technique, which enables a person to skip the uploading process and acquire the possession from the files immediately, when other proprietors of the identical files have submitted these to the cloud server. We prove the safety in our construction, and also the theoretical analysis and experimental results reveal that our construction is efficient used. Within this paper, we introduce the idea of Deduplication dynamic evidence of storage and propose a competent construction known as DeyPoS, to attain dynamic PoS and secure mix-user Deduplication, concurrently. Thinking about the difficulties of structure diversity and tag generation, we exploit a singular tool known as Homomorphic Authenticated Tree (HAT).

Keywords: Cloud storage, dynamic proof of storage, Deduplication.

INTRODUCTION

Users ought to believe that the files kept in the server aren't tampered. A lot of companies, for example Amazon . Com, Google, and Microsoft, provide their very own cloud storage services, where users can upload their files towards the servers, access them from various devices, and share all of them with others. Data integrity is among the most significant qualities whenever a user outsources its files to cloud storage. Traditional approaches for protecting data integrity, for example message authentication codes (MACs) and digital signatures, require users to download all the files in the cloud server for verification, which incurs huge communication cost. They aren't appropriate for cloud storage services [1]. We present additional information about PoS and dynamic PoS. Whenever a verifier wants to determine the integrity of the file, it at random selects some block indexes from the file, and transmits these to the cloud server. Based on these challenged indexes, the cloud server returns the related blocks with their tags. The verifier checks the block integrity and index correctness. However, dynamic PoS cannot encode the block indexes into tags, because the dynamic operations may change many indexes of non-updated blocks, which incurs unnecessary computation and communication cost. Dynamic PoS remains improved inside a multi-user atmosphere, because of the dependence on mix-user Deduplication around the client-side. The previous could be directly guaranteed by cryptographic tags. How to approach the second may be the major distinction between PoS and dynamic PoS. In the majority of the PoS schemes , the block index is “encoded” into its tag, meaning the verifier can look into the block integrity and index correctness concurrently. This signifies that users can skip the uploading process and acquire the possession of files immediately, as lengthy because the submitted files already appear in the cloud server [2]. This method can help to eliminate space for storage for that cloud server, and save transmission bandwidth for users. To the very best of our understanding, there are no dynamic PoS that may support secure mix-user Deduplication. There are two challenges to be able to solve this issue. On a single hands, the authenticated structures utilized in dynamic PoSs, However, even when mix-user Deduplication is achieved , private tag generation continues to be challenging for dynamic operations. In the majority of the existing dynamic PoSs, a tag employed for integrity verification is generated through the secret key from the uploaded. Thus, other proprietors who've the possession from the file but haven't submitted it because of the mix-user Deduplication around the client-side, cannot produce a new tag once they update the file. In cases like this, the dynamic PoSs would fail. For solving private tag generation, each owner can generate its very own authenticated structure and upload the dwelling towards the cloud server, meaning the cloud server stores multiple authenticated structures for every file. The main approaches PoS and dynamic PoS schemes are homomorphic Message Authentication Codes and homomorphic signatures. With the aid of homomorphism, the messages and MACs/signatures during these schemes could be compressed right into a single message along with a single MAC/signature. Therefore, the communication cost could be dramatically reduced. Deduplication during these scenarios would be to deduplicate files among different groups. Regrettably, these schemes cannot support Deduplication because of structure diversity and tag generation. Within this paper, we think about a more general situation that each user features its own files individually. Hence, we concentrate on a Deduplication dynamic PoS plan in multiuser environments.

SYSTEM MODEL

No trivial extension of dynamic PoS is capable of mix-user Deduplication. To fill this void, we present a singular primitive known as Deduplication dynamic evidence of storage. Our body's model views two kinds of entities: the cloud server and users, for every file, original user may be the user who submitted the file towards the cloud server, while subsequent user may be the user who demonstrated the possession from the file but didn't really upload the file towards the cloud server [3]. You will find five phases inside a Deduplication dynamic PoS system: pre-process, upload, Deduplication, update, and evidence of storage. Within the pre-process phase, users plan to upload their local files. The cloud server decides whether these files ought to be submitted. When the upload process is granted, enter in the upload phase otherwise, enter in the Deduplication phase. Within the upload phase, the files to become submitted don't appear in the cloud server. The initial users encode the neighborhood files and upload these to the cloud server. Within the Deduplication phase, the files to become submitted already appear in the cloud server. The following users hold the files in your area and also the cloud server stores the authenticated structures from the files. Subsequent users have to convince the cloud server they own the files without uploading these to the cloud server. Observe that, these 3 phases are performed just once within the existence cycle of the file in the outlook during users. The cloud server and users don't deal with one another. A malicious user may cheat the cloud server by claiming that it features a certain file, however it really doesn't have it or only offers areas of the file. A malicious cloud server may attempt to convince users it faithfully stores files and updates them, whereas the files are broken or otherwise up-to-date. The aim of Deduplication dynamic PoS would be to identify these misbehaviors with overwhelming probability. The Deduplication dynamic PoS are really a comprehensive storage outsourcing approach which establishes mutual confidence between users and also the cloud server inside a multi-user atmosphere. Given personal files, each user that has the whole original file can acquire exactly the same metadata through the initialization formula and pass the Deduplication protocol when the file exists within the cloud server. When a user has submitted the file or passed the Deduplication protocol, it may convince the cloud server that her possession from the file, and could delete the file from the local storage. Regardless of who runs the encoding formula and uploads the encoded file towards the cloud server, the consumer can run the update protocol and also the checking protocol anytime without possessing the file in your area, which signifies our model is appropriate to multi-user environments. Within our model, all users possess the ownerships of the identical file individually, and also the update by one user shouldn't modify the other users. This signifies the cloud server should keep original version and also the new version from the file concurrently once the original file has multiple proprietors [4]. It is possible by using version control techniques that our model can certainly integrate. Uncheatability captures the home of authenticity for mix-user Deduplication around the client-side.

METHODOLOGY

To apply a competent Deduplication dynamic PoS plan, we design a singular authenticated structure known as homomorphic authenticated tree (HAT). A

HAT is really a binary tree by which each leaf node matches an information block. Though HAT doesn't have any limitation on the amount of data blocks, with regard to description simplicity, we think that the amount of data blocks n is equivalent to the amount of leaf nodes inside a full binary tree [5]. Whenever a HAT is initialized, the version quantity of each leaf is 1, and also the version quantity of each non-leaf node is the sum of the those of its two children. We exploit two major algorithms for path search and brother or sister search. The formula takes as input a HAT as well as an purchased listing of the block indexes, and outputs an purchased listing of the node indexes. We define the brother or sister search formula it requires the road ? As input, and outputs the index group of the brothers and sisters of nodes within the path ?. Observe that, the creation of the brother or sister search formula isn't an purchased list. It always outputs the leftmost one out of the rest of the brothers and sisters. Both skip list and Merkle tree would be the classical structures in dynamic PoSs. Since there's no Deduplication plan according to skip list and also the asymptotic performance of skip list is comparable with this of Merkle tree in dynamic PoSs, we simply discuss the Merkle tree within our paper. Merkle tree isn't appropriate for Deduplication in dynamic PoS because of the structure diversity. The purpose of HAT would be to lessen the communication cost in Deduplication. We advise a concrete plan of Deduplication dynamic PoS known as DeyPoS. It includes five algorithms.

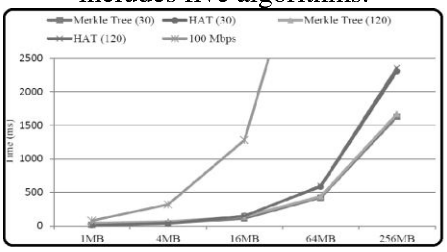

Fig.1.Performance of the proposed system

The evaluation includes three aspects, such as the cost within the upload phase, the price within the Deduplication phase, and also the cost within the evidence of storage phase. The price within the update phase is comparable to the price within the evidence of storage phase, thus, we don't present the price within the update phase. We simply compare our plan using the Merkle tree based solutions. Since there's no Merkle tree based solution that supports both dynamic PoS and Deduplication, we compare our plan using the one according to Merkle tree [6].

CONCLUSION

We developed a novel tool known as HAT which is an excellent authenticated structure. We suggested the excellent needs in multi-user cloud storage systems and introduced the type of Deduplication dynamic PoS. The theoretical and experimental results reveal that our DeyPoS implementation is efficient, particularly when the quality and the amount of the challenged blocks are large. We define the brother or sister search formula it requires the road ? As input, and outputs the index group of the brothers and sisters of nodes within the path ?. Observe that, the creation of the brother or sister search formula isn't an purchased list. It always outputs the leftmost one out of the rest of the brothers and sisters. Both skip list and Merkle tree would be the classical structures in dynamic PoSs. According to HAT, we suggested the very first practical Deduplication dynamic PoS plan known as DeyPoS and demonstrated its peace of mind in the random oracle model. The aim of Deduplication dynamic PoS would be to identify these misbehaviors with overwhelming probability.

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IMPLEMENTING A UNIQUE METHOD USING BILINEAR PAIRINGS FOR OPEN NETS

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Abstract:Our ID-PUIC protocol can also be efficient and versatile. In line with the original client's authorization, the suggested ID-PUIC protocol can realize private remote data integrity checking, delegated remote data integrity checking, and public remote data integrity checking. However, remote data integrity checking can also be an essential security condition in public cloud storage. New security problems need to be solved to be able to help more clients process their data in public places cloud. Once the client is fixed to gain access to Computers, he'll delegate its proxy to process his data and upload them. Increasingly more clients want to store their data to public cloud servers (PCSs) combined with the rapid growth and development of cloud-computing. It can make the clients check whether their outsourced data are stored intact without installing the entire data. In the security problems, we advise a singular proxy-oriented data uploading and remote data integrity checking model in identity-based public key cryptography: identity-based proxy-oriented data uploading and remote data integrity checking in public places cloud (ID-PUIC). We provide the formal definition, system model, and security model. Then, a concrete ID-PUIC protocol was created while using bilinear pairings. The suggested ID-PUIC protocol is provably secure in line with the hardness of computational Diffie-Hellman problem.

Keywords:* *Cloud computing, identity-based cryptography, proxy public key cryptography, remote data integrity checking.

INTRODUCTION

During the last years, cloud-computing satisfies the applying needs and grows very rapidly. Using the public cloud platform, the customers are relieved from the burden for storage management, universal data access with independent geographical locations, etc. Thus, increasingly more clients want to store and process their data using the remote cloud-computing system. Remote data integrity checking is really a primitive that you can use to convince the cloud clients their data are stored intact [1]. In certain special cases, the information owner might be limited to connect to the public cloud server, the information owner will delegate the job of information processing and uploading towards the 3rd party, as an example the proxy. On the other hand, the remote data integrity checking protocol should be efficient to make it appropriate for capacity-limited finish devices. Thus, according to identity-based public cryptography and proxy public key cryptography, we'll study ID-PUIC protocol. Throughout analysis, the manager is going to be limited to connect to the network to be able to guard against collusion. But, the manager's legal business goes on throughout the time of analysis. Whenever a large of information is generated, who are able to help him process these data If these data can't be processed just over time, the manager will face the loss of monetary interest. To avoid the situation happening, the manager needs to delegate the proxy to process its data, for instance, his secretary. But, the manager won't hope others be capable of carry out the remote data integrity checking. Public checking will incur some danger of dripping the privacy. In PKI, the considerable overheads range from heavy certificate verification, certificates generation, delivery, revocation, renewals, etc. In public places cloud-computing, the finish devices might have low computation capacity, for example cell phone, iPod, etc. Identity-based public key cryptography can get rid of the complicated certificate management. To be able to boost the efficiency, identitybased proxy-oriented data uploading and remote data integrity checking is much more attractive [2]. In public places cloud, this paper concentrates on the identity-based proxy-oriented data uploading and remote data integrity checking. By utilizing identity-based public key cryptology, our suggested ID-PUIC protocol is efficient because the certificate management is eliminated. ID-PUIC is really a novel proxy-oriented data uploading and remote data integrity checking model in public places cloud. We provide the formal system model and security model for ID-PUIC protocol. Then, in line with the bilinear pairings, we designed the very first concrete ID-PUIC protocol. Within our suggested ID-PUIC protocol, Original Client will communicate with Computers to determine the remote data integrity. An operating ID-PUIC protocol should be efficient and provably secure. In line with the communication and computation overheads, efficiency analysis could be given. To capture the above mentioned security needs, we formalize the safety meaning of an ID-PUIC protocol.

PROPOSED SYSTEM

We advise a competent ID-PUIC protocol for secure data uploading and storage service in public places clouds. Bilinear pairings technique makes identity-based cryptography practical. Our protocol is made around the bilinear pairings. We first evaluate the bilinear pairings. Then, the concrete ID-PUIC protocol was created in the bilinear pairings. Finally, in line with the computation cost and communication cost, we provide the performance analysis from two aspects: theoretical analysis and prototype implementation. Within the paper, we decide the audience G1 which satisfies the problem that CDH issue is difficult but DDH issue is easy. Around the group G1, DDH issue is easy using the bilinear pairings. (G1, G2) will also be known as GDH (Gap Diffie-Hellman) groups. Around the groups G1 and G2, the fundamental requirement would be that the DLP (Discrete Logarithm Problem) is tough. This concrete ID-PUIC protocol comprises four procedures: Setup, Extract, Proxy-key generation, Tagged, and Proof. To be able to show the intuition in our construction, the concrete protocol's architecture is portrayed. First, Setup is conducted and also the system parameters are generated. In line with the generated system parameters, other procedures are carried out [3]. Within the phase Extract, once the entity's identity is input, KGC generates the entity's private key. Especially, it may create the private keys for that client and also the proxy. Within the phase Tagged, once the data block is input, the proxy generates the block's tag and uploads block-tag pairs to Computers. Within the phase Proxy-key generation, the initial client produces the warrant helping the proxy create the proxy key. Within the phase Proof, the initial client O interacts with Computers. With the interaction, O checks its remote data integrity. First, we provide the computation and communication overhead in our suggested ID-PUIC protocol. Simultaneously, we implement the prototype in our ID-PUIC protocol and evaluate it is time cost. Then, we provide the versatility of remote data integrity checking within the phase Evidence of our ID-PUIC protocol. Finally, we compare our ID-PUIC protocol using the other up-to-date remote data integrity checking protocols. Around the group G1, bilinear pairings, exponentiation, and multiplication lead most computation cost. In contrast to them, another operations are faster, for example, hash function h, the operations on Z^* q and G2, etc [4]. The hash function H can be achieved once for those. Thus, we simply consider bilinear pairings, exponentiation, and multiplication on G1. For that proxy, the computation overhead mainly originates from the phase Tagged. Within the phase Tagged, the proxy performs 2n exponentiation, n multiplication around the group G1, and n hash function h. Within the phase Proof, the initial client O generates the task chalk and Computers reacts to chalk. To be able to show our protocol's practical computation overhead, we've simulated the suggested ID-PUIC protocol by utilizing C programming language with GMP Library and PBC library. National Bureau of Standards and ANSI X9 have determined the shortest key length needs: RSA and DSA are 1024 bits, ECC is 160 bits Based on the standard, and we evaluate our ID-PUIC protocol's communication cost. Following the information systems, the block-tag pairs are submitted to Computers for good. Thus, we simply think about the communication cost that is incurred within the remote data integrity checking. Our suggested ID-PUIC protocol satisfies the non-public checking, delegated checking and public checking. Our contributions will also be appropriate for that scenario of hybrid clouds, in which the proxy may be treatable because the private cloud from the original client. Within the scenario, the non-public cloud will get its private/public key pairs. The non-public cloud will get the proxy-key and also the authorization from the original client with the interaction between your original client and it is private cloud [5]. Once the original client needs its private cloud carry out the data uploading task, it informs its private cloud. Upon finding the original client's instruction, the non-public cloud will communicate with the general public cloud and finished the information uploading task. The safety in our ID-PUIC protocol mainly includes the next parts: correctness, proxy-protection and enforceability.

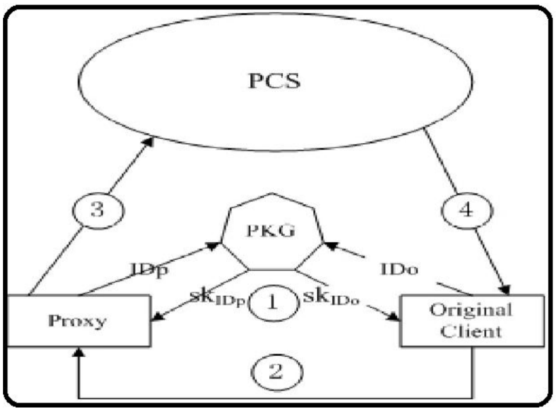

Fig.1.Proposed system

CONCLUSION

Our suggested ID-PUIC protocol satisfies the non-public checking, delegated checking and public checking. Our contributions will also be appropriate for that scenario of hybrid clouds, in which the proxy may be treatable because the private cloud from the original client. Motivated through the application needs, this paper proposes the novel security idea of ID-PUIC in public places cloud. The paper formalizes ID-PUIC's system model and security model. However, the suggested ID-PUIC protocol may also realize private remote data integrity checking, delegated remote data integrity checking and public remote data integrity checking in line with the original client's authorization. Then, the very first concrete ID-PUIC protocol was created using the bilinear pairings technique. The concrete ID-PUIC protocol is probably safe and effective using the formal security proof and efficiency analysis. In PKI, the considerable overheads range from heavy certificate verification, certificates generation, delivery, revocation, renewals, etc. In public places cloud-computing, the finish devices might have low computation capacity, for example cell phone, iPod, etc.

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INVESTIGATING THE SECRET CODE PRIMITIVE IN SECURE SEARCHABLE OPEN DATABASE

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ABSTRACT: In relation to trapdoor generation, as all of the existing schemes don't involve pairing computation, the computation price is reduced in comparison with PEKS generation. Searchable file encryption is of speeding up interest for shielding the information privacy in secure searchable cloud storage. You have to note the trapdoor generation within our plans a little more than individuals of existing schemes because of the additional exponentiation computations. We introduce two games, namely semantic-security against selected keyword attack as well as in distinguish ability against keyword guessing attack1 to capture the safety of PEKS ciphers text and trapdoor, correspondingly. To handle this security vulnerability, we advise a totally new PEKS framework named dual-server PEKS. You have to show a regular construction of secure DS-PEKS from LH-SPHF. During this paper, we investigate security in the well-known cryptographic primitive, namely, public key file encryption with keyword search that's very helpful in a number of applying cloud storage. Regrettably, it has been established the conventional PEKS framework is struggling with an all-natural insecurity known as inside keyword guessing attack (KGA) launched using the malicious server. Since the second primary contribution, we define a totally new variant within the smooth projective hash functions known as straight line and homomorphic SPHF. Including the functionality inside our new framework, we offer a dependable instantiation within the general framework within the Decision Diffie-Hellman-based LH-SPHF and show that could attain the strong security against within the KGA.

Keywords: Keyword search, secure cloud storage, encryption, inside keyword guessing attack, smooth projective hash function, Diffie-Hellman language.

1. INTRODUCTION:

Searchable file encryption may be recognized both in symmetric or uneven files file encryption setting. Precisely, users need to safely share secret keys which you can use for computer file encryption. Otherwise they cannot share the encrypted data outsourced for that cloud. Among the typical solutions may be the searchable file encryption which will help the client to retrieve the encrypted documents which have the client-specified keywords, where because of the keyword trapdoor, the server will uncover the information needed using the user without understanding [1]. To solve this issue, Boneh et al. introduced a much more flexible primitive, namely Public Key File encryption with Keyword Search (PEKS) that allows anyone to look encrypted data within the uneven file encryption setting. Within the PEKS system, when using the receiver’s public key, the sender attaches some encrypted keywords (known as PEKS cipher texts) while using the encrypted data. The receiver then transmits the trapdoor in the to-be-looked keyword for that server for data searching. Because of the trapdoor along with the PEKS cipher text, the server can test once the keyword underlying the PEKS ciphertext is equivalent to the main one selected using the receiver. If that's the problem, the server transmits the matching encrypted data for that receiver. However, the reality is, finish users might not entirely trust the cloud storage servers and might wish to secure their data before uploading individuals towards the cloud server to be able to safeguard the information privacy [2]. No matter being free of secret key distribution, PEKS schemes experience an all-natural insecurity regarding the trapdoor keyword privacy, namely inside Keyword Guessing Attack (KGA). A totally new variant of Smooth Projective Hash Function (SPHF), known as straight line and homomorphic SPHF, is introduced for almost any generic construction of DS-PEKS. We formalize a totally new PEKS framework named Dual-Server Public Key File encryption with Keyword Search (DS-PEKS) to handle safety vulnerability of PEKS. We show a regular construction of DS-PEKS when using the suggested Lin-Hom SPHF.

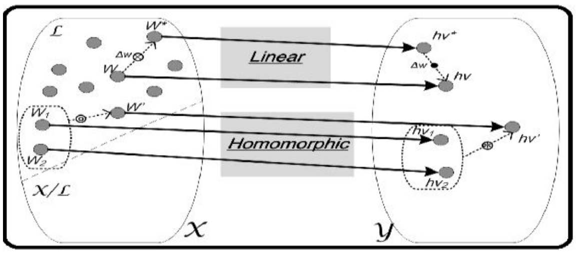

Fig.1.Proposed model

2. PROPOSED MODEL:

To obtain more precise, the KeyGen formula generates the general publicOrpersonal key pairs from the back and front servers instead of this within the receiver. Within the traditional PEKS, since there's just one server, when the trapdoor generation formula is public your server can launch a guessing attack against a keyword cipher text to extract the encrypted keyword. Another one of the conventional PEKS and our suggested DS-PEKS may be the test formula is separated into two algorithms, Front Make certain Back Test operated by two independent servers. This is often required for achieving security from the inside keyword guessing attack. Within the DS-PEKS system, upon acquiring a question inside the receiver, the important thing server pre-processes the trapdoor and PEKS cipher texts getting its private key, then transmits some internal testing-states for that back server while using the corresponding trapdoor and PEKS cipher texts hidden. A corner server will pick which documents are queried using the receiver getting its private key along with the received internal testing-states at the front server [3]. You have to understand that both front server along with the back server here needs to be "honest but curious" and won't collude with one another. More precisely, both servers perform testing strictly transporting out plan procedures but could be thinking about the specific keyword. We must understand that the next security models also imply the safety guarantees outside adversaries that have less capacity in comparison to servers. We introduce two games, namely semantic-security against selected keyword attack and indistinguishability against keyword guessing attack1 to capture the safety of PEKS ciphers text and trapdoor, correspondingly. Semantic-Security against Selected Keyword Attack. The PEKS cipher text doesn't reveal any specifics of the specific keyword for the foe. Indistinguishability against Keyword Guessing Attack. This security model captures the trapdoor reveals no specifics of the specific keyword for that adversarial front server. Adversarial Back Server: The safety types of SS - CKA and IND - KGA in relation to an adversarial back server become individuals against an adversarial front server. Semantic-Security against Selected Keyword Attack. Here the SS - CKA experiment against an adversarial back server is equivalent to the main one against an adversarial front server apart from the foe is supplied the non-public type in the rear server instead of this right in front server. We omit the facts for simplicity. We reference the adversarial back server an internal the SS - CKA experiment just as one SS - CKA foe and define its advantage. Indistinguishability against Keyword Guessing Attack. Similarly, this security model aims to capture the trapdoor doesn't reveal any information for that back server and so is equivalent to that right in front server apart from the foe owns the non-public type in the rear server instead of this right in front server [4]. Therefore, we omit the facts here. Indistinguishability against Keyword Guessing Attack-II. Within our defined security considered IND-KGA-II, it's vital that the malicious back server cannot learn any specifics of the specific two keywords involved in the internal testing-condition. To begin with, we must understand that both keywords involved in the internal-testing condition plays exactly the same role no matter their initial source Therefore, the job within the foe should be to guess the 2 underlying keywords within the internal testing overuse injury in general, rather for each within the initial PEKS cipher text along with the initial trapdoor. Therefore, it's inadequate for the foe to submit number of challenge keywords and so we must hold the foe to submit three different keywords within the challenge stage and guess which two keywords are selected because of the challenge internal-testing condition. A principal component of our construction for dual-server public key file encryption with keyword search is smooth projective hash function (SPHF), an idea created by Cramer and Shoup. During this paper, we must have another critical property of smooth projective hash functions [5]. Precisely, we must hold the SPHF to obtain pseudo-random. During this paper, we introduce a totally new variant of smooth projective hash function. In addition for that original characteristics, we consider two new characteristics- straight line and homomorphic. Our plan's considered because the efficient in relation to PEKS computation. Because our plan doesn't include pairing computation. Particularly, this program necessitates most computation cost because of 2 pairing computation per PEKS generation. In relation to trapdoor generation, as all of the existing schemes don't involve pairing computation, the computation price is reduced in comparison with PEKS generation. Therefore, it may be the computation burden for users who are able to make use of a simple device for searching data. Within our plan, although we have to have another stage for you're testing, our computation price is really lower in comparison with any existing plan once we don't require any pairing computation and searching jobs are handled using the server. You have to note the trapdoor generation within our plans a little more than individuals of existing schemes because of the additional exponentiation computations. You have to understand that this extra pairing computation is carried out across the user side rather within the server.

3. PREVIOUS STUDY:

An affordable solution should be to propose a totally new framework of PEKS. The first PEKS plan without pairings was created by Di Crescenzo and Saraswat. The big event arises from Cock's IBE plan which isn't very practical. The very first PEKS plan needs a secure funnel to supply the trapdoors. To overcome this limitation, Baek et al. suggested a totally new PEKS plan without requiring a great funnel that is actually a good funnel-free PEKS (SCF-PEKS). The concept should be to adding server's public/private key pair in a PEKS system. The keyword cipher text and trapdoor are generated when using the server's public key and so just the server (designated tester) is able to perform search. The first PEKS plan secure against outdoors keyword guessing attacks was suggested by Rhee et al. They enhanced the safety model by presenting the adaptively secure SCF-PEKS, in which a foe is permitted to issue test queries adaptively. Byun et al. introduced the off-line keyword guessing attack against PEKS as keywords are selected within the much smaller sized space than passwords and users usually use well-known keywords for searching documents [6]. The idea of trapdoor indistinguishability was suggested along with the authors proven that trapdoor in distinguish ability could be a sufficient condition to prevent outdoors keyword-guessing attacks.

4. CONCLUSION:

During this paper, we must have another critical property of smooth projective hash functions. During this paper, we suggested a totally new framework, named Dual-Server Public Key File encryption with Keyword Search (DS-PEKS), that may steer obvious from the inside keyword guessing attack that's an natural vulnerability within the traditional PEKS framework. A principal component of our construction for dual-server public key file encryption with keyword search is smooth projective hash function (SPHF), an idea created by Cramer and Shoup. You have to understand that this extra pairing computation is carried out across the user side rather within the server. Therefore, it may be the computation burden for users who are able to make use of a simple device for searching data. We introduced a totally new Smooth Projective Hash Function (SPHF) and attempted round the extender to make a normal DS-PEKS plan. A dependable instantiation within the new SPHF while using Diffie-Hellman problem is also presented within the paper, which gives a dependable DS-PEKS plan without pairings. You have to note the trapdoor generation within our plans a little more than individuals of existing schemes because of the additional exponentiation computations. In relation to trapdoor generation, as all of the existing schemes don't involve pairing computation, the computation price is reduced in comparison with PEKS generation.

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