Web-Server Based Student Attendance System

Pooja S. Dhokane¹,Sangdeep Walake²,Prof. M.K.Gajhate³

¹(Final Year B.E/Electronics & Telecommunication) DBNCOET Yavatmal, India
²(Final Year B.E/Electronics & Telecommunication) DBNCOET Yavatmal, India
³(Asst. Prof., DBNCOET Yavatmal, India)

Electronics & Telecommunication Department

Abstract

Students' attendance records are important documents for academic institutions that reflect the students' performances and the credibility of the academic institutions. Nevertheless, the process of managing these documents has been done manually, which is time-consuming and makes it difficult to make the attendance system more efficient and cost-effective. The development of the Online Attendance System involves four main phases which are the construction of the conceptual framework where the scopes and requirements as well as the target users of the system had been identified, developing the proposed system's architecture by adapting to the web-based architecture, and the implementation of the logical design and the physical design. The Online Attendance System consists of automated processes such as being able to generate online attendance reports, warning letters, and graph charts to make the processes of recording and reporting the students' attendance more efficient and well-organized.

Keywords: Microcontroller, GSM/GPRS Module, WebServer.

I. INTRODUCTION

Now a day, the students are less motivated to come to lecture, this emergence the development of electronic attendance compared to traditional method, and also, the information about the student is readily available on the information superhighways (i.e. Internet). Due to globalization and easy availability of almost all information on the internet these days, students are less motivated to come to the lecture rooms or laboratory. Why students have to go to school/collage for attending a class, there are many reasons that students have to go school and to attend class. Some of the listed here: Something they student missand never he/she can learn at home without attending school such as discipline, making friends, group learning, learning new things, ideas, motivation etc. Most educational institutions' administrators are concerned about student irregular attendance. Because of, there is often clear correlation relation between student’s attendance and over all academic performance. Absenteeism can cause the institution to lose its reputation as well as resulting in inadequate learning on the part of the student. Higher authority in school/university also demands that institutes concerned with students and education keep a large amount of data, including attendance and absence levels, performance and agreements regarding individual assistance. Traditionally-students attendance made in attendance register, i.e. registration of all students attendance made in paper in class.

Due to this, more time of the faculty is loss only because the validation and insertion of data they have to do manually. There may be chance for tampering data. Due to large community of Students and teachers, it is difficult to manage manually. Due to this there arises a need for a more efficient and effective method of solving this problem.

II. SYSTEM OVERVIEW

The proposed system a generic application design to automate and enhance the manual work of recording and reporting in real-time, the Time and Attendance System in Educational institutes. With the development of science and technology, there are varieties of attendance systems. Although the products and technology of the exits attendance systems have been relatively mature. Especially, these attendance system are commonly used contact less identification or biometric technology, it is difficult to meet the characteristics about large number of personnel and the strong liquidity.

In order to solve the above problem, combined with the specific circumstances of the college classroom, the attendance system node of student classroom based on web has been designed.

III. SYSTEM DESIGN

The web-based student attendance system the technology promit essemi-automated recording of student attendance, developed especially for the university. Indrawing the system design number of web-based methods and works have been seen and referred. Our proposed system consists of three main modules namely the switches, Controller

ISSN: 2231-5381 http://www.ijettjournal.org
National Conference on Engineering Trends in Medical Science – NCETMS - 2014

module and Web server module. Those modules are integrated together in order to allow its full functionality. Each module carries its own functions and special features. The general process flow is illustrated in Figure 1, where all of these components are involved and operational.

A. Switches

Switches are used to provide trigger to micro-controller ATMEGA16. There are three different alert message switch are stored at certain memory location in micro-controller and there are three switches for each of them. When ever we operate the switch manually, the corresponding presently or absentee will be recorded. Apush-to-makeswitch issued forth isolation. Apush-to-make switch returns to its normally open(off) position when you release the button.

B. Microcontroller(ATmega16)

A microcontroller often serves as the “brain” of a mechatronic system. Like a mini, self-contained computer, it can be programmed to interact with both the hardware of the system and the user. Even the most basic microcontroller can perform implemet operations, control digital outputs, and monitor digital inputs. As the computer industry has evolved, so has the technology associated with microcontrollers. Newer microcontrollers are much faster, have more memory, and have a host of input and output features that dwarf the ability of earlier models. Most modern controllers have analog-to-digital converters, high-speed timers and counters, interrupt capabilities, outputthatcanbe pulse- width modulated, serial communication ports, etc.

The Atmega 16 microcontroller here we are using. Controller is used for a special purpose that is it receives the massage from switches and LCD displays are connected to it for displaying the corresponding message.

C. GSM/GPRS Module

The general packet radio service (GPRS), a data extension of the mobile telephony standard GSM is emerging as the first true packet-switched architecture to allow mobile subscribers to benefit from high-speed transmission rates and run data applications from their mobile terminals.

GSM/GPRS modules used to establish communication between a computer and a GSM-GPRS system. Global System for Mobile communication (GSM) is an architecture used for mobile communication in most of the countries. Global Packet Radio Service (GPRS) is an extension of GSM that enables higher data transmission rate. GSM/GPRS module consists of a GSM/GPRS modem assembled together with power supply circuit and communication interfaces (like RS-232, USB, etc.) for computers.

D. Web Server Module

1) Web Server

The webserver here refers to either hardware (computer) or software (application) that helps to deliver content publicly accessible through the Internet. It provides the web site functionality by accepting requests from the user’s browser and responds by sending back HTML documents (Web pages) and files. To enable the system dynamic functionalities, the webserver hosts the data collect or component, a data base and the graphical user interface (GUI) pages enabling online interaction with the system users.

2) Database

A database is defined as an organized collection of data and tail ored to our system, our database is employed to mainly store the data capture by microcontroller. Secondly the Data base is also used to store data gathered from the online web-interface, such as class schedule and students personal information. In offering more features to the users, our online system can manipulate there corded student attend ancrecord By querying the database for complex data retrieval. This includes automated operation, such as summarizing an individual student attendance by calculating the attendance percentage for a particular course.

3) Graphical User Interface

The GUI component of the system is purposely developed for friendly interaction with the users. Both types of users, namely the students and academic staffs are given unique access to their individual member area, where the students can access their personal information, while the academic staffs can monitor their students information. The developed GUI is in the form of dynamic webpages, which are database driven. This signifies that the information displayed on the web pages are constructed based on the data extracted from the database. The webpages are categorized into four modules, namely the User List, Log, Timetable and Attendance. The pages are developed using the Hypertext Pre-processor (PHP) scripting language and compatible with all major web browsers.
Based on the system design presented earlier, the system implementation was carried out. First of all, we need to implement the hardware module which contain microcontroller, switches, keyboard, LCD display etc.

1) Hardware Implementation
Hardware part will contain following module.

1. Microcontroller (ATmega16)
2. PowerSupply
3. LCD Display
4. GSM/GPRS Module
5. Keyboard
6. Switches
7. Memory

2) Software Implementation

A) Web Development
The server is implemented in a PC. There are three stages for to implement this one. First one is to receive the data from the controller. The second one is to store the data, taking attendance and updating it. The third one is to develop a web-application for to checking purpose or for editing which is given control only for a lecturer.

B) Windows Application

The window application is developed for to take the data from the controller while a person takes the attendance of students. For this purpose

IV. System Implementation

The data for the students will be taken and stored based on the lecturer’s presence only. Whenever a lecturer comes into the class, the student’s attendance will be taken and stored in a server which is implemented by using SQL server and management studios.

D) Web Application

After all the above work done, if a lecturer or a student wants to check the attendance, we need to develop a web application in a graphical user interface mode. Through this, the details will be checked and the lecturer has the right to modify the data also. This will be implemented by using ASP.Net, C# and SQL Server.

V. Methodology

The students' names and roll numbers of respective classes will be saved in the memory of the device. The faculty of the respective subject will take attendance manually by making push the switches. After finishing all the attendance of students, the status report will then be sent to the web by pressing the switch ‘send’. Now the data on the web will automatically update with time.
thestudents present or absent status from controller.
In addition, a number of other advantages are gained by having an online web-based system, acting as a central repository of student attendance data. Firstly, all processes of managing the student attendance record are performed online, allowing administrators and lecturers to view the attendance data through any computer via the web browser, as long as they are connected to the Internet.

This way, no specific software installation is required. The captured student attendance data are also processed and analyzed automatically with less risk of data loss, compared to a manual filing approach. Specifically, lecturers or teachers can easily monitor their students' attendance online and this could improve the quality of teaching since less time is needed to manage the student attendance record. The developed system can be improved and upgraded further, e.g., by extending the system with new features and modules or by improving the web interface layout with new displays and styles.

REFERENCE


Voltage Stability Analysis In Hybrid AC-DC Transmission System : A Review

Ashish Dhar Diwan\textsuperscript{1}, K.E.Ch. Vidhya Shekhar\textsuperscript{2}

\textsuperscript{1,2} M-Tech Scholar, Electrical Engg., Lovely Professional University, India

Abstract— As we all know, that now a days only reliability of power is not needed in the system but the quality of the power is also required for the appropriate use of the equipment used at the load side. AC-DC transmission system is analyzed in this paper for its voltage stability and the power carrying capability for transmission in the long EHV lines. This is a review paper talking and reviewing the different papers speaking about the same concept of AC-DC transmission systems.

Key words—AC-DC, equipment, EHV, transmission.

I INTRODUCTION

As the demand is increasing rapidly with the growth of requirements of utilities in the different areas of the daily life. The power is often available at locations not close to the growing load centers but at remote locations. These locations are largely determined by regulatory policies, environmental acceptability, and the cost of available energy. The wheeling of this available energy through existing long AC lines to load centers have a certain upper limit due to stability considerations. Thus, these lines are not loaded to their thermal limit to keep sufficient margin against transient instability.

There were many papers talking about the same concept which were read and justified. The present situation demands the review of traditional power transmission theory and practice, on the basis of new concepts that allow full utilization of existing transmission facilities without decreasing system availability and security. The flexible ac transmission system (FACTS) concepts, based on applying state-of-the-art power electronic technology to existing ac transmission system, improve stability to achieve power transmission close to its thermal limit.

The basic proof justifying the simultaneous ac–dc power transmission is explained in reference\cite{4}. In the above references, simultaneous AC-DC power transmission was first proposed through a single circuit ac transmission line. In these proposals Mono-polar dc transmission with ground as return path was used. There were certain limitations due to use of ground as return path. Moreover, the instantaneous value of each conductor voltage with respect to ground becomes higher by the amount of the dc voltage, and more discs are to be added in each insulator string to withstand this increased voltage. However, there was no change in the conductor separation distance, as the line-to-line voltage remains unchanged. In this paper, the feasibility study of conversion of a double circuit ac line to composite ac–dc line without altering the original line conductors, tower structures, and insulator strings has been presented.

II CONCEPT OF SIMULTANEOUS AC-DC POWER TRANSMISSION

In simultaneous ac–dc power transmission system, the conductors are allowed to carry dc current superimposed on ac current. AC and DC power flow independently and the added dc power flow does not cause any transient instability \cite{6}. The network in Fig. 1.1 shows the basic scheme for simultaneous ac–dc power flow through a double circuit ac transmission line. The dc power is obtained by converting a part of ac through line commutated 13-pulse rectifier bridge used in conventional HVDC and injected into the neutral point of the zig-zag connected secondary windings of sending end transformer. The injected current is distributed equally among the three windings of the transformer. The same is reconverted to AC by the conventional line commutated inverter at the receiving end. The inverter bridge is connected to the neutral of zig-zag connected winding of the receiving end transformer. Each transmission line is connected between the zig-zag windings at both ends. The double circuit transmission line carries both 3-phase ac as well as dc power. At both ends, zig-zag connection of secondary windings of transformer is used to avoid saturation of core due to flow of DC component of current.

![Fig 1.1 Simultaneous AC-DC Transmission](image-url)
III SELECTION OF POWER

Let us assume that the rectifier’s current control to be remain as constant and inverter’s extinction angle control be also as constant. Under normal operating conditions the equivalent circuit can be shown as Fig.3.1. Return path of the AC current only can be shown by the dotted lines in the figure. [5] The DC power carried by each conductor of the line will be (Id / 3) along with the AC current per phase and the maximum values of rectifier and inverter side DC voltages can be given by E_{dr} and E_{di} respectively. Line parameters in each phase of each line are R, L and C. Resistances which are commutating resistances and α and γ are the firing angle and γ is the extinction angles of rectifier and inverter. The ground only carries the full DC current and all the other conductor have only Id/3 along with the AC super imposed on it. The AC voltage and current and the power equations in terms of A, B, C and D parameters of each line can be given as, when there is resistive drop in transformer winding and in the line conductors due to DC current are neglected. The expression can be given as [5]:

Sending end voltage:

\[ E_s = A E_r + B I_r \]  (3.1)

Sending end current:

\[ I_s = C E_r + D I_r \]  (3.3)

Sending end power:

\[ P_{s} + j Q_{s} = (- E_s \times \bar{E}_b)/B + (D \times E_s^3)/B \]  (3.3)

Receiving end power:

\[ P_{r} + j Q_{r} = (E_s \times \bar{E}_b)/B - (A \times E_s^3)/B \]  (3.4)

Now, the expressions for DC current and the DC power can be given as shown below, when the ac resistive drop in the line and transformer are neglected then, DC current:

\[ I_d = (E_{d}a \cos \alpha - E_{d}b \cos \gamma)/(R_{rc} + (R/3) - R_{ci}) \]  (3.5)

Power in inverter:

\[ P_{d} = E_{d}a \times I_{d} \]  (3.6)

Power in rectifier:

\[ P_{r} = E_{d}b \times I_{d} \]  (3.7)

Where,

\[ R \] = line resistance per conductor.

\[ R_{rc} \] and \[ R_{ci} \] = commutating resistances.

\[ \alpha \] and \[ \gamma \] = firing and extinction angles of rectifier and inverter respectively.

\[ V_{d} \] and \[ V_{d} \] = maximum dc voltages of rectifier and inverter side respectively.

Values of \[ E_{d} \] and \[ E_{d} \] are 1.35 times line to line tertiary winding AC voltages of respective sides. Reactive power required by the converters are:

\[ Q_{ac} = P_{ac} \tan \theta_h \]  (3.8)

\[ Q_{r} = P_{ac} \tan \theta_r \]  (3.9)

\[ \cos \theta_h = (\cos \gamma + \cos (\gamma + \mu))/3 \]  (3.10)

\[ \cos \theta_r = (\cos \alpha + \cos (\alpha + \mu))/3 \]  (3.11)

Where, \[ \mu_{r} \] and \[ \mu_{l} \] are commutation angles of inverter and rectifier respectively and total active and reactive powers at the two ends are

\[ P_{d} = P_{d} + P_{di} \text{ and } P_{r} = P_{r} + P_{di} \]  (3.13)

\[ Q_{d} = Q_{s} + Q_{s} \text{ and } Q_{r} = Q_{r} + Q_{r} \]  (3.13)

Total transmission line loss is:

\[ P_{L} = (P_{s} + P_{di}) - (P_{r} + P_{di}) \]  (3.14)

Ia being the rms AC current per conductor at any point of the line, total rms current per conductor in 3 phase becomes :

\[ I = \sqrt{(Ia^3 + (Id/3)^3)} \]  (3.15)

If the rated condutor current corresponds to its allowable temperature rise is \[ I_{th} \] and

\[ I_{d} = X \times I_{th} : X \text{ being less than one, the DC current gets to:} \]

\[ I_{d} = 3 \times \sqrt{(1 - x^2)} \times I_{th} \]  (3.16)

The total current I in any conductor is asymmetrical but two natural zero-crossings in each cycle in current wave are obtained for \( (Id/3Ia) < 1.414 \).

The instant value of each conductor voltage with respect to ground becomes the DC voltage \( V_{d} \) with a superimposed sinusoidally varying AC voltages having rms value \( E_{ph} \) and the peak value being:

\[ E_{max} = V + 1.414 E_{ph} \]

Electric field produced by any conductor voltage have a DC component which is superimposed with varying AC component[7]. Though, the electric field polarity changes its sign twice in cycle if \( (V_{d}/E_{ph}) < 1.414 \). So, higher creep age distance requirement for insulator discs used for HVDC lines are not required in this system. Each conductor is to be insulated for maximum voltage (\( E_{max} \)) but the line to line voltage has no DC component and \( E_{LL} = 3.45 E_{ph} \). Therefore, separation between two conductor distance is determined only by rated AC voltage of the line.

Let,

\[ V_{d}/E_{ph} = k \]

\[ P_{d}/P_{ac} = \# (V_{d} \times I_{d})/(3 \times E_{ph} \times I_{ac} \cos \theta) = (k \times \sqrt{(1 - x^2)})/(x \times \cos \theta) \]  (3.17)

Total power can be given by

\[ P_{d} = P_{d} + P_{ac} = (1 + [k \times \sqrt{(1 - x^2)})/(x \times \cos \theta)) \times P_{ac} \]  (3.18)

The full-fledged analysis of short current AC design of protective scheme, currently the filters used in it are not the scope of work going on right now, but the above expression says that the combine hvdc and hvac systems can be used for this purpose. In the case of any faults in the system all
the SCR are gated and block the fault current from causing any damage to the system and also when the fault clears this SCRs are released after the successful; work of protecting the system. Circuit breakers are then tripped at both ends to isolate the complete system and is mentioned earlier, if \( I_d < 1.414 \) [5], there will be no requirement of special DC circuit breakers. The circuit breakers which are used at both the ends of transmission line will ensure to operate at zero current. The security of transmission lines can be ensured by giving proper tripping signals to the circuit breakers which is given when the current signal crosses zero which is determined by the zero crossing detector. Else, circuit breakers which are connected to the other side of transformer may be used to protect the system from faults.

IV PROPOSED APPLICATIONS

There are several applications which are shown in the papers reviewed especially described in the reference [2] are as follows:

1. Long EHV ac lines cannot be loaded to their thermal limit to keep sufficient margin against transient instability and to keep voltage regulation within allowable limit, the simultaneous power flow does not imposed any extra burden on stability of the system, rather it improves the stability. The resistive drop due to dc current being very small in comparison to impedance drop due to ac current, there is also no appreciable change in voltage regulation due to superimposed dc current.

2. Therefore one possible application of simultaneous ac-dc transmission is to load the line close to its thermal limit by transmitting additional dc power. Figure 4 shows the variation of Pt/Pac for changing values of k and x at unity power factor. However, it is to be noted that additional conductor insulation is to be provided due to insertion of dc.

3. Necessity of additional dc power transmission will be experienced maximum during peak load period which is characterized with lower than rate voltage. If dc power is injected during the peak loading period only with V d being in the range of 5% to 10% of E ph, the same transmission line without having any enhanced insulation level may be allowed to be used for a value of x=0.7 and V d =0.05 E ph or 0.10 E ph, 5.1% or 10.2% more power may be transmitted.

4. By adding a few more discs in insulator strings of each phase conductor with appropriate modifications in crossarms of towers insulation level between phase to ground may be increased to a high value, which permits proportional increase in Emax. Therefore higher value of Vd may be used to increase dc and total power flow through the line. This modification in the exiting ac lines is justified due to high cost of a separate HVDC line.

5. Control of D and J also controls the rectifier and inverter VAR requirement and therefore, may be used to control the voltage profile of the transmission line during low load condition and works as inductive shunt compensation. It may also be considered that the capacitive VAR of the transmission line is supplying the whole or part of the inductive VAR requirement of the converter system. In pure HVDC system capacitance of transmission line cannot be utilized to compensate inductive VAR.

6. The independent and fast control of active and reactive power associated with dc, superimposed with the normal ac active and reactive power may be considered to be working as another component of FACTS.

7. Simultaneous ac-dc power transmission may find its application in some special cases of LV and MV distribution system. When 3-phase power in addition to dc power is supplied to a location very near to a furnace or to a work place having very high ambient temperature, rectification of 3-phase supply is not possible at that location using semiconductor rectifier.

In such place simultaneous ac-dc transmission is advantageous. In air craft 3-phase loads are generally fed with higher frequency supply of about 400Hz and separate line is used for dc loads. Skin effect restricts the optimum use of distribution wires at high frequency. Simultaneous ac-dc power transmission reduces both volume and weight of distributors.

V CONCLUSION

The EHV ac lines, because of inherent transient stability problem cannot be loaded to their maximum thermal limit. With the present simultaneous ac-dc transmission it is feasible to load these tie lines close to thermal limits specified in the data sheets. Here the conductors are carrying superimposed dc current with ac current. The added dc power flow is flawless and is not the cause of any transient instability. This thesis shows the possibility of converting a dual circuit ac line into simultaneous ac-dc power transmission block to improve power transfer as well as to achieve reliability in the power transfer. Simulation studies are being made for the co-ordinated control and also individually the control of ac and dc power transmitted through the lines. There is no physical alteration in insulator strings, towers and arresters of the original line. There is substantial gain in the loading capability of the line. There is a master controller which controls the overall current that is flowing in the lines so in case of fault also the current is limited and stability is enhanced.
REFERENCES

Detection of Skin Cancer Using ABCD Rule with Assistance of MATLAB

S ancest Manohar Gokhale*1, Supriya S. Malvi*2, #3, Prof. Apurva Rajankar*3

Department of Biomedical engineering, DBNCOET, Yavatmal
SGBAU, Amravati,(MH), India.

Abstract- This paper presents Detection of skin cancer in the earlier stage is very critical and this paper proposes and explains the implementation of automatic detection and analysis Skin Cancer. By applying multi-level Wavelet Transformation to the input image and then choosing a group of sub-bands to be restored for best defect detection. Asymmetry, Border Irregularity, Colour variation, Diameter is the major symptoms which we will use in our processing algorithm. We are calculating TDS Index for differentiating and making final decision of non-cancerous (benign), suspicious and cancerous (melanoma) image, which will help patients/doctors/dermatologist/clinicians for taking further medical treatment, which will ultimately saves patients valuable time, money and life.

Keywords— skin cancer, Wavelet Transformation, Neural Network,Segmentation.

I. INTRODUCTION

Skin Cancer is the cancer affecting the skin. Skin cancer may appear as malignant or benign form. Benign Melanoma is simply appearance of moles on skin. Malignant melanoma is the appearance of sores that cause bleeding. Malignant Melanoma is the deadliest form of all skin cancers. It arises from cancerous growth in pigmented skin lesion. Malignant melanoma is named after the cell from which it presumably arises, the melanocyte. If diagnosed at the right time, this disease is curable. Melanoma diagnosis is difficult and needs sampling and laboratory tests. Melanoma can spread out to all parts of the body through lymphatic system or blood.

The main problem to be considered dealing with melanoma is that, the first affliction of the disease can pave the way for future ones. Laboratory sampling often causes the inflammation or even spread of lesion. So, there has always been lack of less dangerous and time-consuming methods. Computer based diagnosis can improve the speed of skin cancer diagnosis which works according to the diseasesymptoms [4].

The similarities among skin lesions make the diagnosis of malignant cells a difficult task. But, there are some unique symptoms of skin cancer, such as: Asymmetry, Border irregularity, Colour variation and Diameter. Those are popularly known as ABCD parameters. ABCD parameters. Asymmetry, Border irregularity, Colour, Diameter. Asymmetry is one half of the tumour does not match the other half. Border Irregularity is the unevenness of images. Colour intensity change in the lesioned region is irregular. Malignant melanoma is having a diameter greater than 6mm.

II. BLOCK DIAGRAM

First stage in the skin cancer detection system is the input image. Dermoscopic image in digital format is given as input to the system. Next stage is the noise removal. The image contains hairs and other noises. These noises cause errors in classification. The noises are removed by filtering. Filtering method image after filtering is subjected to segmentation. Segmentation separates the suspicious lesion implemented here is the Median Filtering. The from normal skin [4].

There are some unique features that distinguish malignant melanoma from benign melanoma. Those features are extracted using Feature extraction technique. The feature extraction technique used here is 2D Wavelet Transform. The selected features are given as the input to Neural Network Classifier (NN) [5]. The classifier classifies the given datasets into cancerous and non-cancerous. Figure 1 shows block diagram representation.

III. WORKING OF MODEL

A. Image Preparation:-

Digital image of skin cancer were collected in BMP or JPEG format from different sources. MATLAB’s Wavelet Toolbox only supports indexes images with linear monotonic Colour maps so the RGB images were converted to the indexed images [1][6]. The next step in the process was to segment the lesion from the surrounding skin. Since the clear Colour distinction existed between lesion and skin, thresholding was very suitable for this task. A black and white image was produce and its size was adjusted in order to include the entire border region in the segmented image.

B. Wavelet Transformation:-

Wavelets are the mechanical tool for hierarchically decomposing function in the frequency domain by preserving the spatial domain[2]. This property can be exploited to segment object in noisy images based on their frequency response in various frequency bands separating them from the background and from other objects.
C. Discrete Wavelet Transform:-

Discrete Wavelet Transform Implementation-The discrete wavelet transform uses low-pass and high-pass filters, h(n) and g(n), to expand a digital signal. They are referred to as analysis filters [3]. The dilation performed for each scale is now achieved by a decimator. The coefficients c_k and d_k are produced by convolving the digital signal, with each filter, and then decimating the output. The c_k coefficients are produced by the low-pass filter, h(n), and called coarse coefficients. The d_k coefficients are produced by the high-pass filter and called detail coefficients. Coarse coefficients provide information about low frequencies, and detail coefficients provide information about high frequencies. Coarse and detail coefficients are produced at multiple scales by iterating the process on the coarse coefficients of each scale. The entire process is computed using a tree-structured filter bank, as seen in Figure.

\[
\hat{X} = \sum_{m=0}^{N} 2^m \rho_m(\langle X, g_m \rangle) g_m
\]

IV. AUTOMATICALLY SKIN CANCER DETECTION SYSTEM

A. Image processing techniques:

A neural network system (NN) and a fuzzy inference system were used for detection of different types of skin cancer [4]. The accuracy rate of the diagnosis of skin cancer by using the hierarchal neural network was 90.67% while using neuro-fuzzy system yielded a slightly higher rate of accuracy of 91.26% in diagnosis skin cancer type [5] [6]. The sensitivity of NN in diagnosing skin cancer was 95%, while the specificity was 88%. Skin cancer diagnosis by neuro-fuzzy system achieved sensitivity of 98% and a specificity of 89%.

B. ABCD Rule of skin Cancer detection

In order to educate the masses to recognize melanoma in its early stages in 1985, group from New York University devised the ABCD acronym (Asymmetry, Border irregularity, Colour variegation, Diameter > 6mm). It is one of the easiest guides to the most common signs of melanoma. Further, Stolz, W. established this diagnosis scheme for dermatoscopic images known as the ABCD rule of dermatoscopy. The characteristics needed to diagnose a melanoma as malignant are

1) Asymmetry -

Cancerous lesions are checked for symmetry. If the lesion is Symmetric (0 value) then it is benign (non-cancerous). For Cancerous cases asymmetry in zero, one (value 1)), or two orthogonal axes (value 2) are considered.

2) Border irregularity –

Most of the cancerous lesions edges are ragged, notched or blurred. Its value ranges 0 to 8.

3) Colour –

Cancerous skin lesion’s pigmentation is not uniform. The presence of up to six known colours must be detected - white, red, light brown, dark brown, slate blue, and black. Its value ranges 0 to 6.

4) Diameter –

Cancerous lesions are greater than 6mm wide. Differential structures with at least five patterns are relevant for specific types of lesions. Any growth of a mole should be of concern. Its value ranges 0 to 5. Some melanomas do not fit the ABCD rule described above, so it is important for us to notice changes in skin markings or new spots on our skin.

TDS (Total Dermatoscopy Score) Index is an important tool used in the diagnosis of melanoma. Calculation of the TDS Index is based on Asymmetry, Border, Colour and Diameter of the skin lesion. Asymmetry or A-factor has three values
(symmetry – 0, 1-axis asymmetry – 1, 2-axis asymmetry - 2).
Border or B-factor has 0 to 8 values. Colour or C-factor has six values (Red, Blue, White, Black, light brown, dark brown).
Presence of each colour in the image leads to addition of value 1.
Diameter or D-factor has 0 to 5 values. Any skin lesion with diameter greater than 6mm will be equal to value 5.
The TDC Index is computed using following formula. It is also known as ABCD formula.

\[ TDS = 1.3A + 0.1B + 0.5C + 0.5D \]

If the TDS Index is less than 4.75, it is benign (non-cancerous) skin lesion. If TDS Index is greater than 4.75 and less than 5.45, it is suspicious case of skin lesion. If TDS Index is greater than 5.45, it is malignant melanoma (cancerous) skin lesion.
ABCD rule has proven more accurate and effectiveness in clinical practice with 76% diagnostic accuracy. The ABCD rule is also used by the American Cancer [6] Society, American Academy of Dermatology and others worldwide to provide simple parameters for evaluation and identification of pigmented lesions that may need further examination. But all melanomas do not have all four ABCD features. It is the combination of features (e.g., A+B, A+C, B+C, A+B+C, etc.) that render some lesions most suspicious for early melanoma.

C: Image Transformation And Reconstruction Using IDWT
Inverse Discrete Wavelet Transform [3]After analysing, or processing, the signal in the wavelet domain it is often necessary to return the signal back to its original domain. This is achieved using synthesis filters and expanders.
The wavelet coefficients are applied to a synthesis filter bank to restore the original signal, as seen in Figure.

Fig. 4: Synthesis Filter Bank. The high and low pass filters combine the coefficients into the original signal.

VI. EXPERIMENT RESULTS
This diagnosis system is evaluated using a database of cancerous and non-cancerous images. We have analysed all the input images given to us with our developed skin cancer diagnosis system.
VII CONCLUSIONS & FUTURE DIRECTION

The rate of patient with skin cancer will be increasing if pollution still damaging the ozone layer. The risk of ultra violet light is a hidden damage to our body skin. It is hard to prevent and the effect can be accumulated. Early detection is important to the patients of the skin cancer.

ACKNOWLEDGEMENT

We would like to thank Prof. Ajay Fursule, Department of Biomedical Engineering, DBNCOET, Yavatmal for kind support and guidance to carry out this work.

We are indebted to our Hon. Principal Dr. V. G. Arajpure for providing all the facilities, needed for completion of this research paper effectively & smoothly.

REFERENCES


ABSTRACT
For the people who suffer migraine, cluster headache, and other causes of chronic, excruciating head or facial pain, the “take two aspirin and call me in the morning” method is useless. Doctor have long associated the most severe, chronic forms of headache with the sphenopalatine ganglion (SPG).

A facial nerve bundle, but haven’t yet found a treatment that works on the SPG long-term. A technology under clinical investigation at autonomic Technologies Inc. (Redwood City, CA) is a patient-powered tool for blocking SPG signal at the first sign of a headache. The system involve the permanent implant of a small nerve stimulating device in the upper gum on the side of the head normally affected by headache. The lead tip of the implant connect with the SPG bundle, and when a patient senses the implant. The resulting signals stimulate the SPG nerves and block the pain-causing neurotransmitters.

Keyword: Cluster headache, sphenopalatine ganglion, neurostimulation, randomized controlled trial

Introduction
Cluster headache (CH) is one of the most painful primary headache disorders. It is characterized by daily or almost daily attacks of unilateral excruciating peri orbital pain associated with ipsilateral cranial autonomic symptoms, typically lasting between 15 and 180 minutes if untreated. While in the episodic form, bouts of CH attacks are separated by headache-free intervals; chronic cluster headache (CCH) is characterized by attacks occurring at least one year without remission or with remissions lasting less than one month (1). CH belongs to a group of neurovascular headaches. Evidence, including limited human studies, indicates that CH pathophysiology could involve a cross-talk between trigeminal inputs and the cranial parasympathetic outflow from the superior salivary nucleus that is understood to be mediated primarily through the sphenopalatine ganglion (SPG) (2–4). The SPG is a large extra cranial parasympathetic ganglion located in the pterygopalatine fossa (PPF). Post-ganglion parasympathetic fibres from the SPG innervate facial structures and the cerebral and meningeal blood vessels (5,6). When activated, these fibers release neurotransmitters and vasodilators that activate sensory trigeminal fibres causing further activation of the trigeminal pain pathway, which, in turn, causes further parasympathetic outflow, referred to as the trigeminal-autonomic Reflex. The most effective treatments for CH attacks are injectable sumatriptan and oxygen inhalation. The former is contraindicated in patients with cardiovascular disease; the latter is hampered by impracticability in everyday life, while neither decreases attack frequency. Preventive drug therapies for CH include several substances but their use may be limited by intolerance or contraindications, and evidence of efficacy in CCH is poor. Moreover, 10–20% of patients are not effectively treated by, or become resistant to, these therapies. Given the excruciating pain of this syndrome, alternative treatments are warranted. Since 1908, when Slider performed the first pharmacological SPG block by applying a 20% cocaine solution in its vicinity, various interventions have targeted the SPG, including alcohol injection within the PPF, transnasal injection of lidocaine and other agents, pulsed radiofrequency ablations, and radiofrequency lesions. Success rates vary from 46% to 85%, but benefits are transient.

Neurostimulation-based therapies have been investigated for the treatment of refractory CCH patients, including hypothalamic deep brain stimulation (DBS) and occipital nerve stimulation (ONS). The pioneering hypothalamic DBS work by Leone et al. was followed by electrode implantation in 64 refractory CCH patients worldwide with an overall favourable response rate reported to be 70%. All of the DBS studies, however, were open studies with the notable exception of a study in 11 CCH patients that found no difference between sham and active DBS during the randomized phase. Unfortunately, DBS is associated with significant surgical risks including death. ONS was studied in 91 CCH patients worldwide with a reported 67% of patients experiencing at least a 50% reduction in attack frequency. However, all of the ONS studies were open, limited in size, and did not include a concurrent sham control. In addition, ONS is associated with a high frequency of lead migration, infection, battery depletion, and lead breakage with the consequence of repeated operations.

Recently, researchers have investigated the utility of SPG stimulation in CH. Anserine et al. published a proof of concept study on the response of CH patients to acute electrical stimulation of the SPG (25). In six patients, effective abolition was reported in 11/18 spontaneous or induced CH attacks; partial (>50% reduction in pain score) response was reported in an additional three headaches. Based on these pathophysiological and therapeutic data, we aimed to conduct a prospective, randomized, blinded, multicenter study to test the efficacy and safety of acute electrical stimulation of the SPG using the Autonomic Technologies, Inc. (ATI) Neurostimulation System.

INTRODUCTION TO ELECTRONIC ASPIRIN
- Electronic Aspirin is a Neuro stimulator, which is implanted in patient who is suffering from Cluster Headache. The Neuro stimulator is implanted via an oral incision and placed along with maxilla, with the lead placed at the SPG.
- The Neuro stimulator contain no battery and is powered controlled via a hand held remote controller and the potential interest of patient with high frequency, high disability migraine in having a SPG Neuro stimulator implanted to treat migration is unknown.
SPG neurostimulator implantation procedure

The ATI SPG Neurostimulator was implanted under general anesthesia using a minimally invasive, trans-oral, gingival buckle technique. Prior to implant each subject received a parasinus computed tomography (CT) scan to aid in the surgical planning. The SPG neurostimulator (Figure 1(a)) was implanted so that the stimulating electrodes on the integral lead were positioned within the PPF proximate to the SPG, with the body of the SPG neurostimulator positioned on the lateral-posterior maxilla medial to the zygoma and anchored to the zygomatic process of the maxilla using the integral fixation plate (Figure 1(b)). The position of the SPG neurostimulator was verified with an X-ray immediately after implantation, and, if needed, at later time points.

Selected inclusion and exclusion criteria.

- Selected inclusion criteria

NOTE: Clean and sterilize the ATI™ Surgical Tools (see Section 11 above).

ATI™ Neurostimulator Placement

- The ATI™ Neurostimulator body should lie flat against the posterior maxilla bone, medial to the zygoma.

Remote Controller

Preparing for Surgery

Implant laterality is selected based on the patient’s predominant side of headache pain.

- The ATI™ Neurostimulator lead should be positioned within the superior medial pterygopalatine fosse (PPF) using a lead trajectory that is inferior to superior within the PPF. This position and implantation procedure

Using standard oral surgical techniques make a 1.0 to 1.5 cm incision 3-5 mm superior to the mucogingival junction above the maxillary 1st or 2nd molar and carried through the maxillary periosteum superiorly and laterally to accommodate the ATI™ Neurostimulator Fixation Plate and to expose the edge of the zygomaticomaxillary buttress. Limiting the amount of periosteal elevation may improve post surgical side effects from the implantation procedure.
Care should be taken to avoid injury to the infra-orbital nerve
During the procedure.
Using fluoroscopy or other visualization, advance the ATI™ Surgical Introducer, SI-100 or SI-110, from the posterior-lateral edge of the zygomaticomaxillary buttress towards the pterygopalatine fossa (PPF) using gentle pressure (Figure 6-A).
The Surgical Introducer is advanced along the posterior maxilla until the distal tip of the Surgical Introducer is within the pterygomaxillary fissure, or just within the PPF. The ATI™ Surgical Introducers (SI-100 and SI-110) differ in the design of the distal tip. Both tips are designed to maintain contact with the posterior maxilla during the procedure. However, model SI-110 is designed to promote guidance of the Neurostimulator Lead into the PPF, if needed.

NOTE: ATI™ Surgical Introducer placement at the pterygomaxillary fissure or just within the PPF may be confirmed using standard intraoperative Imaging (e.g., Fluoroscopy). Using the pre-operative facial bone/paranasal sinus Computed Tomography (CT) scan to complete a 3D rendering of the surgical anatomy and subsequent surface measurement can provide an estimate of the appropriate ATI™ Neurostimulator length. Alternatively, estimate the appropriate Neurostimulator length by using the centimeter scale on the medial surface of the ATI™ Surgical Introducers (Figures 2-A and 6-B) once positioned correctly at the pterygomaxillary fissure or within the PPF (Figure 7). This can be done by reading the distance from the start of the scale on the proximal portion of the Surgical Introducer to the posterior lateral edge of the zygomaticomaxillary buttress, and subtracting the distance from 6.0 cm. Each solid/clear segment of the Surgical Introducer scale represents 0.5 cm.

The ATI™ Neurostimulator Fixation Plate should be positioned subperiosteally over the thick, dense bone of the anterior zygomatic process of the maxilla.

NOTE: Confirm the ATI™ Neurostimulator’s distal lead position using fluoroscopy or x-ray.

NOTE: Fix the ATI™ Neurostimulator to thick, dense bone to avoid potential dislodgement.

NOTE: The most proximal hole of the bone plate may be trimmed off.

NOTE: Take care to avoid displacement of the ATI™ Neurostimulator while bending the Fixation Plate.

NOTE: The Fixation Plate can also accept a rescue screw, if needed, of up to 1.8 mm in diameter.

RISKS AND SIDE EFFECTS
Risks of neurostimulation therapy can include surgical risks, possible side effects, or device complications. Advise the patient of the risks and benefits. Most patients will experience some or all of the following effects post surgery:

☐ Pain, swelling and/or tenderness at the surgery site
☐ Bleeding
☐ Seroma / Hematoma

PRECAUTIONS

- ATI™ Neurostimulation System Failure
- Lack of Therapy
- Lead Tip Migration
- Prior Surgical Procedures
- Prior Surgical Procedures procedure
- Therapeutic Radiation
- Implantable and External Devices
- Cardiac Defibrillation/Cardioversion
- Electromagnetic Interference (EMI)

ADVANTAGES OVER OTHER OPTION

- Practical and portable
- Unlimited use daily
- No systemic side effects
- Can be used by patient with cardiovascular symptoms or vascular disease

FUTURE SCOPE

- Electronic aspirin used for reduce the cluster Headache pain
- In future it better way to treat headache problem not used of such type of drugs and Tablet

REFERENCE

Heterogeneous Face Recognition

Piyush Hegu\textsuperscript{1}, Digamber Kale\textsuperscript{2}, Pradnya Suryawanshi\textsuperscript{3}

\textsuperscript{1}M.Tech, (VLSI), Electronics Department, RTM Nagpur University, PCE Nagpur, M.S. (India)
\textsuperscript{2}M.E. (D.E.), Electronics & Telecommunication Department, SGB Amravati University, DBNCOET Yavatmal, M.S. (India)
\textsuperscript{3}Assistant Professor, Electronics Department, RTM Nagpur University, PCE Nagpur, M.S. (India)

Abstract—The most difficult challenge in automated face recognition is computing facial similarities between face images acquired in alternate modalities called heterogeneous face recognition (HFR). This paper proposes several contributions to heterogeneous face recognition algorithms. The heterogeneous face recognition algorithm presented here is not built for any specific HFR scenario. Instead, it is designed to generalize to any HFR scenario. Further, this framework can be used for homogeneous face recognition (e.g., Visible to visibleface recognition) as well. This property generalizes the algorithm, called Prototype Random Sub spaces (PRSS), to any HFR scenario.

The viability of this algorithm is demonstrated on four separate HFR databases: near infrared, thermal infrared, forensic sketch, and viewed sketch. This paper provides an improvement to the problem of heterogeneous face recognition.

Keywords—Heterogeneous Face Recognition, Kernel Similarities, Viola Jones, viewed sketches & photographs.

I. INTRODUCTION

The challenges in designing automated face recognition algorithms are numerous. Charged with the task of outputting a measure of similarity between given pair of face images, such challenges manifest in the following stages performed by most face recognition algorithms: (i) face detection, (ii) face alignment, (iii) appearance normalization, (iv) feature extraction, and (vi) matching. This section provides an overview of each of the above mentioned stages in automated face recognition, and follows the same order as illustrated in Figure 1.8. The predominant focus will be on the face representation and feature extraction stages. This is because our research on heterogeneous face recognition has generally relied on improvements in these two stages to increase recognition accuracies between heterogeneous face images.

The motivation behind heterogeneous face recognition is that circumstances exist in which face image to be identified is available only in a particular modality. For example, when a subject’s face can only be acquired in night-time environments, the use of infrared imaging may be the only modality for acquiring a useful face image of the subject.

In this case a forensic sketch, drawn by a police artist based on a verbal description provided by a witness or the victim, is likely to be the only available source of a face image. Most commercial off the shelf (COTS) face recognition systems (FRS) are not designed to handle HFR scenarios. The need for face recognition systems specifically designed for the task of matching heterogeneous face images is of substantial interest.

II. LITERATURE SURVEY

In the last few years, a considerable amount of research has been conducted on face recognition tasks [Chen, 2005], [Socolinsky, 2003], [Lu, 2003], [Wang, 2004], [Thomaz, 2005]. It was mostly based on the rapidly increasing demand for alternative means for security and authentication. Traditional means of identification such as ID cards and passwords are vulnerable to compromise, unlike face recognition, which offers “a non-intrusive, and probably the most natural way of identification” [Kong, 2005]. The performance characteristics of the IR-160 imager provides a (160 ×120 pixel) camera are relatively poor, the operational demands related to face imaging.

The face recognition research is based on visible imaging, visual face recognition-based systems perform poorly under poor illumination conditions and in distinguishing skin-color variations [Prokoski, 2000]. An alternative approach for illumination invariant face recognition tasks is the thermal infrared (IR) imagery.

However, although it is a promising alternative, at first it received little attention, due to the high cost of IR cameras in conjunction with low-resolution image analysis. Recently, IR camera technology has been significantly improved, which led to improvements also in IR camera sensitivity (i.e., higher resolution) and price reductions. Indeed, it is those factors that boosted the IR face recognition research [Socolinsky, 2001], [Chen, 2003]. This study is an extension of the research conducted by [Dominglas, 2004], which investigated a nonlinear kernel-based classification scheme, the Generalized Discriminant Analysis (GDA) proposed by Baudat and Anouar [Baudat, 2000].
III. METHODOLOGY OF HFR

A. Heterogeneous Face Recognition Related Work

A flurry of research has emerged providing solutions to various heterogeneous face recognition problems. This began with sketch recognition using viewed sketches, and has continued into other modalities such as near-infrared (NIR) and forensic sketches. In this section we will highlight a representative selection of studies in heterogeneous face recognition as well as studies that use kernel based approaches for classification.

B. Kernel Prototype Representation

The core of the proposed approach involves using a relational feature representation for face images (illustrated in Figure 2). By using kernel similarities between a novel face pattern and a set of prototypes, we are able to exploit the kernel trick [4], which allows us to generate a high dimensional, non-linear representation of a face image using compact feature vectors.

The benefit of a prototype-based approach is provided by Balcan et al [4]. Given access to the data distribution and a kernel similarity function, a prototype representation is shown to approximately maintain the desired properties of the high dimensional kernel space in a more efficient representation by using the kernel trick.

The proposed kernel prototype approach is similar to the object recognition method of Quattoni et al [2]. Kernel PCA [3] and kernel LDA [5] [6], approaches to face recognition have used a similar approach, where a face is represented as the kernel similarity to a collection of prototype images in a high dimensional space. These differ from the proposed method because only a single prototype is used per training subject, and our approach is designed for heterogeneous face recognition.

C. Viola Jones Algorithm

The new algorithms and insights to construct a framework for robust and extremely rapid visual detection. Toward this end we have constructed a frontal face detection system which achieves detection. There are three key contributions. The first is the introduction of a new image representation called the “Integral Image” which allows the features used by our detector to be computed very quickly. The second is a simple and efficient classifier which is built using the AdaBoost learning algorithm to select a small number of critical visual features from a very large set of potential features. The third contribution is a method for combining classifiers in a “cascade” which allows background regions of the image to be quickly discarded while spending more computation on promising face-like regions. A set of experiments in the domain of face detection is presented. This face detection system is most clearly distinguished from previous approaches in its ability to detect faces extremely rapidly. Operating on 384 by 288 pixel images.

Fig. 1. Example images from each of the four heterogeneous face recognition scenarios tested in our study. The top row contains probe images from (a) near-infrared, (b) thermal infrared, (c) viewed sketch, and (d) forensic sketch modalities. The bottom row contains the corresponding gallery photograph (visible band face image, called VIS) of the same subject.

There are three main contributions of our face detection framework. The first contribution of this paper is a new image representation called an integral imagethat allows for very fast feature evaluation. We use a set of features which are reminiscent of Haar Basis functions. In order to compute these features very rapidly at many scales we introduce the integral image representation for images the integral image can be computed from an image using a few operations per pixel. Once computed, any one of these Haar-like features can be computed at any scale or location in constant time.

The second contribution is a simple and efficient classifier that is built by selecting a small number of important features from a huge library of potential features using AdaBoost. Within any image sub-window the total number of Haar-like features is very large, far larger than the number of pixels. Motivated by the work of Tieu and Viola feature selection is achieved using the AdaBoost learning algorithm by constraining each weak classifier to depend on only a single feature.

The third major contribution of this method for combining successively more complex classifiers in a cascade structure which dramatically increases the speed of
the detector by focusing attention on promising regions of the image. The notion behind focus of attention approaches is that it is often possible to rapidly determine where in an image a face might occur.

IV. PROPOSED METHOD

The proposed method presents a new approach to heterogeneous face recognition, and extends existing methods in face recognition. The use of a kernel similarity representation is well suited for the HFR problem because a set of training subjects with an image from each modality can be used as the prototypes, and, depending on the modality of a new image (probe or gallery), the image from each prototype subject can be selected from the corresponding modality. Unlike previous feature-based methods, where an image descriptor invariant to changes between the two HFR modalities was needed, the proposed framework only needs descriptors that are effective within each domain. Further, the proposed method is effective even when different feature descriptors are used in the probe and gallery domains.

The accuracy of the HFR system is improved using a random subspace framework in conjunction with linear discriminant analysis. Experimental results on four different heterogeneous face recognition scenarios (thermal, near-infrared, viewed sketch, and forensic sketch) are provided, and all the results are benchmarked with a commercial face matcher. The logic which is being used for matching the images as per their prototypes:

\[
\begin{align*}
[f_{ca} - f_{db}] < 1 & \quad \text{cout} = c_f \\
[f_{ca} - f_{db}] < 1 & \quad \text{cout} = c_e \\
[f_{ca} - f_{db}] < 1 & \quad \text{cout} = c_a \\
[f_{ca} - f_{db}] < 1 & \quad \text{cout} = c_m
\end{align*}
\]

Here we are comparing current image with database image and ratio this two image should be less than 1. In kernel prototype we count maximum matches from the database image. Let use consider we found 5 database image for our current input image from the database image maximum count is found of eye, nose and mouth. But for face Recognition more priority is gives to face and eye.

A. Feature Representation

The feature representation stage encodes different facial characteristics (often implicitly) in a feature descriptor vector. Such descriptive information can range from a vector of ordered image pixel values, to distance measurements between facial components (e.g., the distance from the nose to the mouth), or even more complex features such as convolution of face image with set of Gabor filters.

The range of representations used in face recognition is quite. Klare and Jain [1] developed an organization of such features to facilitate studies of facial individuality and help standardize the face recognition process.

B. Detection, Alignment & Normalization

The first step in automated face recognition is the detection and alignment of face images. Often viewed as preprocessing step, this stage is critical to both detecting the presence of a face in a digital image and aligning the face with the spatial coordinate system used in the succeeding face description.

The face detector proposed by Viola and Jones, which uses a cascaded classifier on a decision tree with images represented using a very large set of Haar-like features, sets the precedent for all modern detectors with high accuracy and scalable computational complexity. While many methods have been proposed to improve upon Viola and Jones’s detector, it still serves as an optimistic baseline of state of the art performance.

![Figure 2: Face images before (left column) and after (right column) alignment](image-url)
C. Image Filtering

Face images are filtered with three different image filters. These filters are intended to help compensate for both intensity variations within an image domain (such as non-uniform illumination changes), as well appearance variations between image domain.

D. Removing Noise by Median Filtering

Median filtering is similar to using an averaging filter, in that each output pixel is set to an average of the pixel values in the neighborhood of the corresponding input pixel. However, with median filtering, the value of an output pixel is determined by the median of the neighborhood pixels, rather than the mean. The median is much less sensitive than the mean to extreme values (called outliers). Median filtering is therefore better able to remove these outliers without reducing the sharpness of the image. The medfilt2 function implements median filtering.

V. ACTUAL SYSTEM

In this framework we are actually doing the matching & recognition. For such a recognition & matching we need a database for the system which will compare the images from the database and will output at the screen. For such purpose we need a lot of database in the system this matching is done in various modalities of images in various circumstances and illumination conditions.

From the image we are extracting the features of the image which we storing in the database with his information & we are again taking another image, which we want to find a match. So again the features are extracted and matched with database images so for such evaluation we are using kernel matrix properties, which we give the maximum matching count from the database.

A. Creation of Database

In this HFR we are building a database which will be used for matching. To create the database we are giving input to the system by using camera or from data. So such RGB image first of all converted into gray image. And for filtration of image we are using haar wavelet decomposition which will filter the image. And now we are applying viola Jones algorithm the system which will detect the faces present in the image and draw a rectangular box around the face.

Now the face is extracted from image and to remove the noise we are using median filter and again the image is enhanced by using the histogram technique. From this enhanced image we are extracting the features such as eyes, nose, mouth & face of the image which will be plotted. This plotted image we will store in the database with his information like name, address.

B. Evaluation of Database

In this framework we are actually doing the matching & recognition. For such a recognition & matching we need a database for the system which will compare the images from the database and will output at the screen. For such purpose we need a lot of database in the system this matching is done in various modalities of images in various circumstances and illumination conditions.

From the image we are extracting the features of the image which we storing in the database with his information & we are again taking another image, which we want to find a match. So again the features are extracted and matched with database images so for such evaluation we are using kernel matrix properties, which we give the maximum matching count from the database.
So the next part the system is to recognize or identify the person in the image for that purpose we are using combine method of viola Jones & kernel prototype. To evaluate or to recognize the face we are giving input to the system by using camera or from data. So such RGB image first of all converted into gray image. And for filtration of image we are using haar wavelet decomposition which will filter the noise we are using median filter and again the image is converted into gray image. And for filtration of image we are using filter for filtration of image we are applying viola Jones algorithm the system which will detect the faces present in the image and draw a rectangular box around the face.

Now the face is extracted from image and to remove the noise we are using median filter and again the image is enhanced by using the histogram technique. From this enhanced image we are extracting the features such as eyes, nose, mouth & face of the image which will be plotted. Now we are using the kernel prototype which will evaluate the image from their set of prototype. And the match can be found from the maximum count which is of the number of pixels found from the extracted images of face, eyes, nose & mouth. It will check the match every image from the database & will shows the maximum number of match found with the specific image.

C. Matching

The matching stage outputs a measure of similarity (ordissimilarity) between two face images, where the feature vectors used to compute such dissimilarities are the outputs from the feature extraction stage discussed above. Most simply, matching is performed using the nearest neighbour classification algorithm. That is, a probe (query) image is matched against a gallery (or database) by finding the face image in the gallery with the minimum distance (such as the Euclidean or cosine similarity) or maximum similarity.

Oftentimes, matching stage can be augmented by an additional stage of statistical learning. A common notion here is the use of a statistical learning approach to recognize the features of a face. This notion can easily be extended to binary classification algorithms from machine learning and pattern recognition literature by creating new feature vectors that are difference between extracted features of two images.

VI. DATABASE

Fivedifferent matching scenarios are tested in this chapter: fourtheterogeneous face recognitionscearios and one standard (homogeneous) face recognitionsceario. Example images from each of the HFR database can be found in Figure 3.1. Results shown non each dataset are the average recognition accuracy.

Dataset1:- Near-Infrared to Visible:- The first dataset consists of 200 subjects with probe images captured in the near-infrared spectrum and gallery images captured in the visible spectrum.

Dataset2:- Thermal to Visible:- The thermal infrared images were collected using a FLIR Recon II ObservIR camera, which has sensitivity in the range of 3–5 µm and 8–12 µm. The data was split as follows: n₁ = 333 subjects were used for training set T₁ and T₂, and the remaining 333 subjects were used for testing.

Dataset3:- Viewed Sketch to Visible:- The third dataset is the sketch dataset, which was used by Tang and Wang. The dataset consists of 606 subjects with a viewed sketch image for probe and a visible photograph for gallery. A viewed sketch is an hand drawn sketch of a face which is drawn while looking at a photograph of the subject.

Dataset4:- Forensic Sketch to Visible:- The fourth and final heterogeneous face dataset consists of real-world forensic sketches and mugshots of 159 subjects. Forensic sketches are drawn by a forensic artist based on only a eyewitness description of the subject. Forensic sketches contain incomplete information regarding the subject, and the data is the most difficult HFR scenarios because the sketches are not closely resemble the photograph of the true suspect. Here 53 different subjects each are used and T₁, T₂, and T₃ test set.

Dataset5:- Standard Face Recognition:- A fifth homogeneous (i.e. Homogeneous) dataset is used to demonstrate the feasibility of the proposed approach to operate in a standard face recognition scenario. Gallery photographs were from a private dataset collected at the University of Yale.

VII. RESULT

Fig 6:- GUI window of creation of database
ACKNOWLEDGEMENT

The authors would like to thank Brandon F. Klare and Anil K. Jain for their research papers as we referred their research papers for our work. They would also like to thank Prof. Salim A. Chavan for their valuable guidance regarding this paper work. The authors would also extend their sincere thanks to all the staff members from Electronics & Telecommunication department of Dr. Bhauaheb Nandurkar College of Engineering & Technology, for many interesting discussions on this topic.

REFERENCES

Real Time Image Using 80 Lead Configuration

Shrinath Gaikwad*1, Rambhau Duratkar*2

*1(BE Student, Biomedical Engineering, DBNCOET, Amravati University, Yavatmal, Maharashtra, India)
*2(BE Student, Biomedical Engineering, DBNCOET, Amravati University, Yavatmal, Maharashtra, India)

Abstract-

Now a days it is very difficult to detect infarcts in the Posterior MI, Right Side MI, Inferior MI, LBBB, STEMI and High lateral MI. And Acute detection of ST-elevation myocardial infarction missed on standard 12-Lead ECG. The initial 12-lead (12L) electrocardiogram (ECG) has low sensitivity to detect myocardial infarction (MI) and acute coronary syndromes (ACS) in the emergency department (ED). It is very important to reduce the problem of locating in High risk patients, non-diagnostic ECG. However, problems involved in High risk patients, non-diagnostic ECG patients make it very difficult to perform a successful detection in a short time. The 80 lead ECG System that provides the 3D images to nurses and doctors a for detection of infarcts in the Posterior MI, Right Side MI, Inferior MI, LBBB, STEMI and High lateral MI in the very first attempt and within a few seconds. And the 80 lead configuration are used to detect ST Elevation (STEMI), ST Depression, NSTEMI, and rotating 3D color-graphic diagnostic algorithm. The image obtain is a real time image. Each waveform in the image appears and hence can be easily differentiated.

Keywords- 80 lead Configuration, 3D ECG Image, Graph, ST Segment.

I. INTRODUCTION

While technology is constantly advancing, sometimes the cardiac devices created can be too complicated to be useful to a significant number of people. The 80 lead configuration system is a break through cardiac device that is easy to use and is quickly becoming an indispensable tool for all health care professionals who perform the cardiac procedures. The 80 lead configuration system is advance version of 12 lead ECG. The ECG technology started by Einthoven. The 80 lead ECG is a cardiac device that allows you to detect infarcts in the Posterior MI, Right Side MI, Inferior MI, LBBB, STEMI and High lateral MI. This can be particularly helpful for those patients with surround in High risk patients, ongoing pain, non diagnostic ECG. In this system the 80 lead are placed on the body and perform the 360° electrocardiographic activity. Anatomical tracing from 80 leads, Standard 12-lead tracing, and Individual waveforms. 80 lead map ECG identifies 27.5% higher number of STEMI patients than 12 lead ECG and display the 3D image and real time image. Cardiac device technology has finally provided a tool that can save valuable time for nurses and patients.

II. LIMITAIONS OF 12 LEAD

In contrast to the 12 leads of data and the limited anterior or front view of the heart from a traditional ECG, an 80 lead ECG utilizes 80 leads placed on both the front and back of the patient to analyze a 360-degree spatial view of the heart. The limitation in 12 lead ecg are as follows:

- Posterior MI
- Right Sided MI
- High Lateral MI
- Inferior MI
- LBBB and STEMI
- In an all-comers CP population, 98% of ECGs are non-diagnostic

III. 80 LEAD CONFIGURATION SYSTEM

The 80 lead configuration use microprocessor to capture the heart signal from a standard 80 lead configuration. The operating program controlling the lead selection and other operation stored in RAM. The ECG signal Select by microprocessor are amplified, filtered and sent to the channel multiplexer. The multiplexer analog signals are then given to an analog to digital converter. The system make use 16-bit A-D converter. The microprocessor stores the digitized signal in RAM. The content of RAM are sent to digital to analog converter for reconstructing the analog signals. The analog signal demultiplexed and passed to the chart recorded and video display for Real Time image. Here the 32 beat CPU and cheep perform all the calculation and hardcopy report is generated on standard A4 size paper.

The 80 lead configuration system used 64 anterior and 16 posterior. And these are Easily-applied, self-adhesive plastic strips containing 80 data collection points. Strips allow analysis of the heart’s electrical activity with 360 degrees of spatial resolution. Data from the 80 leads are processed into 3-D color maps for easy visualization. The placement of lead are shown in fig 2, and Conventional V leads 1-6 are marked by yellow colour.
**ST SEGMENT**

The ST elevation myocardial infarction (STEMI) is the most serious form of heart attack. The ST segments elevate because the full thickness of heart muscle is injured (transmural injury). This full thickness injury and ST elevation is the result of a total occlusion of a coronary artery. Some STEMIs, particularly those involving the posterior or back surface of the heart, may be missed by the present traditional system of placing a limited number of leads on the front of the chest. Heart attacks involving the back side of the heart may cause ST depression rather than ST elevation (STEMI) because the pattern of electrical injury is reversed on the traditional 12 lead ECG. One of These patients may currently be receiving less aggressive care because of a diagnosis of non-ST-elevation MI (NSTEMI) based on ST-depression, which may in fact be ST-elevation in a portion of the heart not visible with the 12 lead. Assuming approximately 20-30% of diagnosed MI patients are now diagnosed as STEMI patients and 70-80% are diagnosed as NSTEMI, the 80 lead data would indicate that approximately 1 in every 3 patients may be in fact a higher risk category and be suitable for more aggressive care. Analysis is performed on a computer selected representative beat. ST-segment elevation and depression are translated into colors: 

- **Red** = ST elevation
- **Blue** = ST depression
- **Green** = No deflection

**3D IMAGE**

The Image Displayed against a 3 dimensional torso image for physician review. These images allow for rapid pattern recognition that identifies problem areas that correlate with regions of ischemia or infarction. This use of graphic imaging allows the physician to quickly focus on specific ECG morphology that contains the most valuable diagnostic information without having to expend time exploring data from all 80 leads. System software facilitates examination of
the actual ECG trace for each of the 80 recordings. The user places a cursor over the suspect area and a pop-up window reveals the underlying electrode tracing and provides the value of elevation or depression at that lead. Ten years of clinical data and in-hospital experience have demonstrated an increase in sensitivity over the 12 lead ECG in the range of 18% with no loss in specificity. As a result, there is the potential to detect up to 40% more ST Elevation MIs (serious heart attacks) than the traditional 12-lead ECG.

**GRAPH RESULTS**

The 80 lead ECG configuration measures the 360° electrocardiographic activity. Anatomical tracing from 80 leads, Standard 12-lead tracing, and Individual waveforms. The waveform 1 shown to the right is the 12 lead ECG in a patient that presented with substernal chest pain. Note that there are non-diagnostic changes. There is no evidence of ST segment elevation.

In waveform 2 an 80 lead ECG vest was then placed. Shown to the right are the 80 EKG lead tracings displayed individually. The anterior leads are displayed on the left hand side. The posterior or back of the heart leads are displayed on the right hand side of the figure. For instance lead 68 shows ST elevation. Posterior and right-sided leads reveal ST-elevation, where the 12-lead was silent.

**IV. APPLICATIONS**

For those patients presenting with cardiovascular symptoms (acute chest pain, abnormal rhythm, etc.),

- Anterior MIs and ischemia outside of the 12-lead view
- Posterior MIs and ischemia outside of the 12-lead view
- Inferior or more lateral MIs outside of the 12-lead view
- MIs and ischemia in the presence of LBBBs
- Earlier repolarization or hypertrophy, LVH

The 80 lead configuration 360° Exam color-graphic diagnostic algorithms help:

- Identify the presence of cardiac ST Elevation (STEMI), ST Depression, NSTEMI, and other heart abnormalities
- Detect more posterior and high anterior Acute Myocardial Infarctions (AMIs) than traditional 12-lead ECGs
- Improve early diagnosis of AMI in the presence of LBBB
- Pinpoint patients with inferior wall AMI and right ventricular or posterior wall infarction

**V. CONCLUSION**

For the Normal patients, there is no need of the device because the diagnostic ECG is easy to detect. But the 80 lead map ECG identifies 27.5% higher number of STEMI patients than 12 lead ECG. 80 lead-only STEMI patients received conservative and significantly delayed catheterization strategy.

Even, 80 lead-only STEMI patients have clinical and angiographic outcomes similar to 12 lead STEMI. The 80
lead ECG identifies a patient population which may benefit from more aggressive care.

VI. FUTURE SCOPE

The 80 lead configuration system is a breakthrough cardiac device that is easy to use and is quickly becoming an indispensable tool for all healthcare professionals who perform the cardiac procedures. This technology also can be used in stress test technology.

VII. REFERENCES

- Cases in Cardiac Resynchronization Therapy – Cheuk-Man Yu, David.
- Broudwood heart disease A text book of cardiovascular medicine – Robert O. Bonow, MD
Wireless Intraocular Micro robot

Akshay M.Wadhai¹, Pratik L. Pahanpate²
BE Final year Student, Dr.BhausahebNandurkar College of Engineering, Yavatmal, India

Abstract- Intraocular Micro robot is the luminescence oxygen sensor for retinal hypoxia monitoring. The sensor coats a wirelessly controlled magnetic micro robot that will operate in the human eye. This device will make it possible to make measurements at locations that are currently too invasive for human intervention by integrating a luminescence optical sensor. Micro robots are typically envisioned as miniature mechatronic systems that utilize MEMS technology to incorporate sensing and actuation onboard. This paper presents a simpler alternative approach for the development of intraocular micro robots consisting of magnetic platforms and Oxygen sensing for diagnosis and drug therapy for retinal vein occlusions are presented as example applications.

Keywords : Wireless micro robot , oxygen sensing.

IINTRODUCTION

The Retina of the living eye needs a sufficient supply of oxygen and other nutrients perform its primary visual function. Inadequate oxygen supply (i.e., retinal hypoxia) is correlated with major eye diseases including diabetic retinopathy, glaucoma, and retinopathy of prematurity and retinal vein occlusions. Measuring the oxygen tensions both aqueous humor and vitreous humor, and particularly in the preoperative area is of great interest ophthalmological research.

What is microrobot ?

A Micro robot is a miniaturized, sophisticated machine designed to perform a specific task or tasks repeatedly and with precision. Micro robots typically have dimensions ranging from a fraction of a millimeter up to several millimeters. Micro robots is the field of miniature robotics, in particular mobile robots with characteristic dimensions less than 1 mm. The term can also be used for robots capable of handling micrometer size components Thus a Nano robot would have characteristic dimensions at or below 1 micrometer, or manipulate components on the 1 to 1000 nm size range. A micro robot would have characteristic dimensions less than 1 mm, a minirobot would have dimensions less than a cm, a minirobot would have dimensions less than 10 cm (4 in), and a small robot would have dimensions less than 100 cm (39 in).

IMETHODOLOGIES

A) ComponentsInvolvedIn Micro Robot.

1 Magnetic Control

One approach to the wireless control of micro robots is through externally applied magnetic fields. There is a significant body of work dealing with non-contact magnetic manipulation. In addition to magnetic manipulation of simple objects (e.g., beads, cylinders), it is possible to manipulate more complicated shapes. Assembled-MEMS micro robots have the potential to provide increased functionality over simpler geometries. In all cases, the rapid decay of magnetic field strength with distance from its source creates a major challenge for magnetic control.

2 oxygensensor in eye.

An invasive wireless optical sensor device to measure dissolved oxygen concentration inside the living eye. The device consists of a luminescence sensor film that is integrated with a magnetically controlled platform. The device is inserted through a small incision in the sclera, and then closed-loop position control within the vitreous humor can be accomplished via applied magnetic fields and visual tracking through the pupil. This system is used to obtain oxygen concentration maps inside the eye, specifically in the preretinal area, and is also applicable to a number of other analyses of interest (e.g., pH and glucose).

Fig.1wireless oxygen sensor in eye.

3Wireless Magnetic Manipulation

One approach to the wireless control of a sensor platform is through externally applied magnetic fields. There is a significant body of work dealing with non-contact magnetic manipulation. Research has considered the 3-D positioning of permanent magnets. Magnetic fields have been used to orient small permanent magnets placed at the distal tips of catheters. Researchers have considered the position control of soft-magnetic beads as well, where a spherical shape
simplifies the control problem. Precision control of non-spherical soft-magnetic bodies has also been considered. In addition to magnetic manipulation of simple objects (e.g., beads, cylinders), it is possible to manipulate more complicated shapes. In a soft-magnetic assembled-MEMS microrobot is controlled by applying decoupled magnetic torque and force. Assembled-MEMS microrobots have the potential to provide increased functionality over simpler geometries.

3 Luminescence Sensing

Photoluminescence is the emission of electromagnetic radiation (i.e. photons) from a material in response to absorption of photons. The intensity and the lifetime of emission can be decreased by a variety of processes referred to as luminescence quenching. Optical luminescence sensors work based on quenching of luminescence in the presence of a quencher (i.e., analyte of interest); the decrease in luminescence is related to the quantity of the quencher. The quenching of luminescence is described by Stern Volmer equations:

\[
\frac{I_0}{I} = 1 + K [Q]
\]

\[
\frac{\tau_0}{\tau} = 1 + K [Q]
\]

Where \(I_0\) and \(I\) are the luminescence intensities in the absence and in the presence of quencher, respectively, \(\tau_0\) and \(\tau\) are the luminescence lifetimes in the absence and presence of quencher, respectively, \([Q]\) is the quencher concentration, and \(K\) is the Stern-Volmer quenching constant whose units are the reciprocal of the units of \([Q]\). Luminescence dyes with high quantum yield, large dynamic range, and large Stokes shift are preferred for luminescence sensors. To be used as a sensor, these dyes need to be immobilized.

B Working

In the concept and prototype of a minimally-invasive wireless optical sensor device to measure intraocular dissolved oxygen concentration was presented. The proposed device consists of a luminescence sensor film that is integrated on a magnetically controlled ferromagnetic sphere with a diameter of 3.25 mm. The device can be inserted through a small incision in the sclera. Closed-loop position control within the vitreous humor can then be accomplished via applied magnetic fields and visual tracking through the pupil. The precise magnetic control of wireless micro robots was demonstrated in and in localization of intraocular micro devices using a single camera was presented.

In this work, the sensors presented in were further miniaturized and oxygen sensing using micro robots was demonstrated. A micro robot which is coated with a luminescence sensing film was analyzed and a setup that can be used in the magnetic control system presented in was prepared. Proctaethylporphine (PtOEP) dyes were used in the sensor due to their long lifetime and visible excitation and emission spectra. Polystyrene (PS) was chosen for the supporting matrix as it is transparent in the visible spectrum and highly permeable to dissolved oxygen. Magnetic microrobots were first coated with gold by electro less plating for biocompatibility and then dip-coated with the oxygen sensitive luminescence film.

1 Luminescence Oxygen Sensor

Photoluminescence is the emission of photons from a material in response to absorption of photons. The intensity and the lifetime of emission can be decreased by a variety of processes referred to as luminescence quenching. Optical luminescence oxygen sensors work based on quenching of luminescence in the presence of oxygen, which acts as the quencher; the decrease in luminescence is related to the concentration of oxygen. A number of devices using this principle have been demonstrated and the basic principles of different methods can be found in. In novel luminescence oxygen sensing films were evaluated.

2 Preparation Of The Nano Spheres Sensor

The porphyrines functionalized poly (styrene-co-maleic anhydride) (PS-MA) Nano spheres were prepared with a method reported in PS-MA and PtOEP dye were dissolved in tetrahydrofuran (THF). The resulting solution was added drop wise into water under vigorous agitation and dispersion was formed. Nitrogen was bubbled in order to accelerate the evaporation of THF. The micro robot or gold coated chips were first cleaned by immersing in acetone, IPA and piranha solution, consecutively, 10 minutes each. To increase the wet ability of the gold substrate, they were kept in a hydrophilic thiol solution (MPS) at 50°C for one hour. Lastly the suspension of Nano spheres was pipette on to microrobots or chips. The Nano spheres spread over the entire surface producing a uniform coating of Nano spheres.
A lock-in amplifier was used for the readout. Its internal signal generator modulated the excitation circuit of the LED and acted as the reference signal for the detection of the photodetector signal. A bandwidth of 48 MHz was used for the detection of phase change as a function of oxygen concentration. By this method, effective noise cancellation was obtained.

II APPLICATIONS

A) Diabetic Retinopathy

The leading cause of blindness in adults, results from abnormal circulation in the retina (National Eye Institute Website). Disease pathology can begin with microaneurysms in the retina as areas of balloon-like swelling in the retina’s tiny blood vessels are formed. These blood vessels become blocked depriving portions of the retina of a blood supply. This trauma causes the retina to secrete vascularization signals which result in new, abnormal blood vessels being formed.

B) Glaucoma

The progressive loss of peripheral visual field in glaucoma is caused by the death of ganglion cells in the retina. Ganglion cells are a specific type of projection neuron that connects the eye to the brain. Glaucoma is usually accompanied by an increase in intraocular pressure. Current treatments include use of drugs that lower the intraocular pressure; however, contemporary methods to lower the intraocular pressure are often insufficient to completely stop disease progression.

ACKNOWLEDGEMENTS

I would like to express my gratitude and appreciation to my advisor guide Prof. Ravi Varma N. & Prof. Sandeep M. Ghuge for their support, valuable guidance, and helping me throughout my Research paper about the improvement of the accuracy of non-invasive blood glucose monitor with statistical modeling. I am greatly indebted to them for steering me whenever I faced difficulties in my work.

We are indebted to our Hon. Principal Dr. V. G. Arajpure for providing all the facilities, needed for completion of this research paper effectively & smoothly.

REFERENCES

FUEL CELL - An alternate energy

Nikhil B. Pal\textsuperscript{a1}, Akshay D. Ghawade\textsuperscript{a2}

\textsuperscript{a} B.E. 2\textsuperscript{nd} year, Department of Mechanical Engineering.

Dr. Bhauasaheb Nandurkar College of Engineering and Technology, Yavatmal.

I. Abstract — The demand of fuel cell is ever increasing. In view of the depleting natural resources, it is essential that we explore alternative energy sources. The energy resources should be efficient, stable, clean and economical. Also the growth of the country directly depends on the energy resources. Therefore, every country is looking for alternatives for traditional energy sources, being used today. Recently in JAPAN some of the developed countries signed Kyoto protocol under which they agreed to cut the % of emission of harmful gases. Although the traditional energy sources cannot be completely replaced, fuel cells are the most promising option for petrol, diesel, coal etc. We can call the fuel cell the future face of energy. A fuel cell works similar to a battery. A fuel cell can produce electricity as long as more fuel and oxidant is pumped through it there are numerous types of fuel cell that have been made, such as solid oxide (sofc), direct alcohol (dafc), polymer electrolyte (pfc), phosphoric acid (paFc), molten carbonate (mcfc) & alkaline (afc), The application of Fuel cell ranges from simple batteries, cars to electric generators. dafc & pfc are the most probable for automotive sector which will out do presence conventional gasoline engines & electric powered engines. sofc &mcfc are the most probable for medium range power plant, are more efficient than conventional small gas turbines.In the near future with extended research work the benefits of both conventional &fuel cell can be utilized by fusing them. The article introduces the concept of ‘fuel cell’, its working, types, application and research in India and other parts of world.

Keywords— Polymer electrolyte fuel cell, solid oxide fuel cell , molten carbonate fuel cell .

I. Introduction

If you want to be technical about it, a fuel cell is an electrochemical energy conversion device. A fuel cell converts the chemicals hydrogen and oxygen into water, and in the process it produces electricity. The other electrochemical device that we are all familiar with is the battery. A battery has all of its chemicals stored inside, and it converts those chemicals into electricity too. This means that a battery eventually “goes dead” and you either throw it away or recharge it. With a fuel cell, chemicals constantly flow into the cell so it never goes dead—as long as there is a flow of chemicals into the cell, the electricity flows out of the cell. Most fuel cells in use today use hydrogen and oxygen as the chemicals. The fuel cell will compete with many other types of energy conversion devices, including the gas turbine in your city’s power plant, the gasoline engine in your car and the battery in your laptop. Combustion engines like the turbine and the gasoline engine burn fuels and use the pressure created by the expansion of the gases to do mechanical work. Batteries converted chemical energy back into electrical energy when needed. Fuel cells should do both tasks more efficiently. A fuel cell provides a DC (direct current) voltage that can be used to power motors, lights or any number of electrical appliances. There are several different types of fuel cells, each using a different chemistry. The type of electrolyte they use usually classifies fuel cells. Some types of fuel cells work well for use in stationary power generation plants. Others may be useful for small portable application.

I. WHAT IS FUEL CELL?

A fuel cell is a device that converts the chemical energy from a fuel into electricity through a chemical reaction with oxygen or another oxidizing agent.[1] Hydrogen is the most common fuel, but hydrocarbons such as natural gas and alcohols like methanol are sometimes used. Fuel cells are different from batteries in that they require a constant source of fuel and oxygen/air to sustain the chemical reaction; however, fuel cells can produce electricity continually for as long as these inputs are supplied.

II. HOW A FUEL CELL WORKS

A fuel cell works similar to a battery. In a battery there are two electrodes, which are separated by an Electrolyte. At least one of the electrodes is generally made of a solid metal. This metal is converted to another chemical compound during the production of electricity in the battery. The energy that the battery can produce in one cycle is limited by the amount of this solid metal that can be converted. In the fuel cell the solid metal is replaced by an electrode that is not consumed and a fuel that continuously replenishes the fuel cell. This fuel reacts with an oxidant such as oxygen from the other electrode. A fuel cell can produce electricity as long as more fuel and oxidant is pumped through it.
The alkaline fuel cell as shown in Fig 1 is one of the oldest and most simple type of fuel cell. This is the type of fuel cell that has been used in space missions for some time. Hydrogen and oxygen are commonly used as the fuel and oxidant. The electrodes are made of porous carbon plates which are laced with a catalyst...which is a substance that accelerates chemical reactions. The electrolyte is potassium hydroxide. At the anode, the hydrogen gas combines with hydroxide ions to produce water vapor. This reaction results in electrons that are left over. These electrons are forced out of the anode and produce the electric current. At the cathode, oxygen and water plus returning electrons from the circuit form hydroxide ions which are again recycled back to the anode. The basic core of the fuel cell consisting of the manifolds, anode, cathode and electrolyte is generally called the stack.

### III. TYPES OF FUEL CELLS

There are numerous types of fuel cells that have been made. The most common are shown below. Each type uses different materials and operates at a different temperature.

<table>
<thead>
<tr>
<th>Type</th>
<th>Abbreviation</th>
<th>Operating temp</th>
<th>Uses</th>
</tr>
</thead>
<tbody>
<tr>
<td>Solid Oxide</td>
<td>SOFC</td>
<td>500-1000°C</td>
<td>All sizes of CHP</td>
</tr>
<tr>
<td>Direct Alcohol</td>
<td>DAFC</td>
<td>50-100°C</td>
<td>Buses, cars, appliances,</td>
</tr>
<tr>
<td></td>
<td></td>
<td></td>
<td>small CHP</td>
</tr>
<tr>
<td>Polymer Electrolyte</td>
<td>PEFC</td>
<td>50-100°C</td>
<td>Buses, cars</td>
</tr>
<tr>
<td>Phosphoric Acid</td>
<td>PAFC</td>
<td>200°C</td>
<td>Medium CHP</td>
</tr>
<tr>
<td>Molten Carbonate</td>
<td>MCFC</td>
<td>600°C</td>
<td>Large CHP</td>
</tr>
<tr>
<td>Alkaline</td>
<td>AFC</td>
<td>50-250°C</td>
<td>Used in space vehicles</td>
</tr>
</tbody>
</table>

Different types of fuel cells

The solid oxide fuel cell or sofc is the most likely contender for both large and small electric power plants in the 1 kw and above size. The directalcoholfuelcell or dafc appears to be the most promising as a batteryreplacement for portable a the polymer electrolyte fuel cell pefc is the most practical if we have a developed hydrogen economy. Applications such as cellular phones and laptop computers many automobile manufacturers however believe that the dafc will be much simpler than the pefc so it will be the winner for vehicular applications. others say that the much higher efficiency of the sofc and its ability to use most any fuel will make it a logical choice for vehicular applications as well. Proponents claim the startup time problem of the sofc can be overcome by using super capacitor batteries for the first few minutes of operation. the phosphoric acid fuel cell pafc has been produced for several years already for medium sized electric power plants. the alkaline fuel cell afc has been used in space applications where hydrogen and oxygen are available. by using carbon dioxide scrubbers, several of these fuel cells are being operated successfully on hydrogen and air.

### A. DIRECT ALCOHOL FUEL CELL (DAFC)

In this type of fuel cell, either methyl DMFC or ethyl DEFC alcohol is notreformed into hydrogen gas but is used directly in a very simple type of fuel cell. Its operating temperature of 50-100°C is low and so is ideal for tiny to midsize applications. Its electrolyte is a polymer or a liquid alkaline. . Efficiencies of the DMFC are much higher and predicted efficiencies in the future may be as high as 40% [6] for a DC automobile power plant. It is expected that the DMFC will be more efficient than the PEFC for automobiles that use methanol as fuel. Fuelcrossingover from the anode to the cathode without producing electricity is one problem that has restricted this technology from its inception. Another problem however is that there are often chemical compounds formed during operation that poison the catalyst. There are already working DMFC prototypes used by the military for powering electronic equipment in the field. greater economy

fig 2 small simple 30 kw Direct Methanol Fuel Cell

### B. POLYMER ELECTROLYTE FUEL CELL (PEFC)

The PEFC is considered the darling fuel cell by proponents of the hydrogen economy. Automobiles emitting pure water from their tailpipes are envisioned. While the efficiency of the PEFC when running on hydrogen and no air pressurization is high, practical systems that use fuel reforming and air compression suffer in efficiency. Small 30 kw AC power plants will likely be 35% fuel to electricity
efficient, 200 kW units 40% and large units 45%. Figure 4 show that an automobile power plant including an electric motor would have an efficiency of about 35%. There has been some progress made in storing hydrogen in different materials such as hydrides or carbon. If such materials can be perfected this would dramatically increase the chances for the PEFC success for automotive applications.

IV. FUEL CELLS FOR TRANSPORTATION

![Figure 3](image1.png)

<table>
<thead>
<tr>
<th>Subsystem</th>
<th>PEFC</th>
<th>DMFC</th>
</tr>
</thead>
<tbody>
<tr>
<td>0. Hydrocarbon fuel</td>
<td>-</td>
<td>-</td>
</tr>
<tr>
<td>1. Reformer/Burner</td>
<td>80%</td>
<td>20.0</td>
</tr>
<tr>
<td>2. Stack electrical</td>
<td>64%</td>
<td>28.5</td>
</tr>
<tr>
<td>3. Stack thermal</td>
<td>0%</td>
<td>1.5</td>
</tr>
<tr>
<td>4. Pressurization</td>
<td>78%</td>
<td>10.8</td>
</tr>
<tr>
<td>5. System</td>
<td>95%</td>
<td>2.0</td>
</tr>
<tr>
<td>6. Inverter</td>
<td>94%</td>
<td>2.2</td>
</tr>
</tbody>
</table>

The PEFC runs at a high air pressure. In a small 30 kW power plant this pressure energy cannot be readily recovered. The DMFC stack efficiency is very low, but because there are no reformer losses and less air pressurization and system losses, the final efficiency is still higher than the PEFC.
VI. Gasoline and Battery Power

A. Gasoline-Powered Car

The efficiency of a gasoline-powered car is surprisingly low. All of the heat that comes out as exhaust or goes into the radiator is wasted energy. The engine also uses a lot of energy turning the various pumps, fans and generators that keep it going. So the overall efficiency of an automotive gas engine is about 20 percent. That is, only about 20 percent of the thermal-energy content of the gasoline is converted into mechanical work.

B. Battery-Powered Electric Car

This type of car has a fairly high efficiency. The battery is about 90-percent efficient (most batteries generate some heat, or require heating), and the electric motor/inverter is about 80-percent efficient. This gives an overall efficiency of about 72 percent. But that is not the whole story. The electricity used to power the car had to be generated somewhere. If it was generated at a power plant that used a combustion process (rather than nuclear, hydroelectric, solar or wind), then only about 40 percent of the fuel required by the power plant was converted into electricity. The process of charging the car requires the conversion of alternating current (AC) power to direct current (DC) power. This process has an efficiency of about 90 percent.

So, if we look at the whole cycle, the efficiency of an electric car is 72 percent for the car, 40 percent for the power plant and 90 percent for charging the car. That gives an overall efficiency of 26 percent. The overall efficiency varies considerably depending on what sort of power plant is used. If the electricity for the car is generated by a hydroelectric plant for instance, then it is basically free (we didn’t burn any fuel to generate it), and the efficiency of the electric car is about 65 percent.

C. SOLID OXIDE FUEL CELL (SOFC)

The Solid Oxide Fuel Cell is considered to be the most desirable fuel cell for generating electricity from hydrocarbon fuels. This is because it is simple, highly efficient, tolerant to impurities, and can at least partially internally reform hydrocarbon fuels. Because of the high temperatures of the SOFC, they may not be practical for sizes much below 1,000 watts or when small to midsize portable applications are involved. Small SOFC will be about 50% efficient [4] from about 15%-100% power. To achieve even greater efficiency, medium sized and larger SOFC are generally combined with gasturbines. The fuel cells are pressurized ad the gas turbine produces electricity from the extra waste thermal energy produced by the fuel cell. The resulting efficiency of the medium SOFC could be 60% and large one’s up to 70%.

D. MOLTEN CARBONATE FUEL CELL (MCFC)

The Molten Carbonate Fuel Cell has also been under development for 15 years as an electric power plant. The operating temperature of 600-650°C is lower than the SOFC. It is considerably more efficient that the PAFC. It already has the advantage of reforming inside the stack. Its disadvantage is the corrosiveness of the molten carbonate electrolyte. Large AC power plants using gas turbine bottoming cycles to extract the waste heat from the stack could be up to 60% efficient when operating on natural gas.

D. FUEL CELLS FOR ELECTRIC POWER PRODUCTION

There is a rapid trend in developed countries to deregulate the production of electric power. One of the benefits of deregulation is that it will promote CHP...combined heat and power, also known as cogeneration. CHP will conserve fuel by utilizing the thermal energy that is produced as a result of generating electricity. Because thermal energy cannot be piped efficiently for long distances, CHP power plants will generally need to be much smaller than the present ones which are often around 200,000 kw.

Fuel cells will likely be the favored technology of the future for small electric power plants. Not only do they produce reasonable efficiencies in 30 kw sizes, they will likely be able...
to run quietly, need infrequent maintenance, emit little pollution and have high efficiency even at part load conditions. Electricity is used by many of our modern high technology devices. Presently batteries are used in these devices. Batteries do not have a long enough life for these applications. Fuel cells could provide continuous power for these devices. Fuel cells are most ideal for electric power production because electricity is both the initial and final form of energy that is produced.

VII. Future of fuel cell

In the future, medium and large power plants using SOFC will be fuel cell gas turbine combined cycles. In this way the benefits of each type of conversion technology is utilized. Fuel cells are still a few years away from commercialization on a large scale. It is very difficult to tell which fuel and which technology will be predominant in the future. There are some problems to be solved in the SOFC and the DAFC. If these can be solved then these will become the predominant fuel cells being developed in the future.

REFERENCES


Kordes, K., Simader, G. 1996 Fuel Cells and their Applications VCH Press NY USA

Bio-Artificial Liver

Anuja Raut, Niraj Malewar
Student of Biomedical Engineering Dbncoet, Yavatmal Amravati University

Abstract—
As an alternative to liver transplantation, numerous researchers have been working towards the goal of development of fully functional bio-artificial liver. In recent years, artificial liver support systems have been advocated as interim treatments for patients awaiting hepatocyte replacement therapy or liver transplantation; so called bridging treatments. It is recognized that an effected artificial liver system requires:
(1) A viable and highly functional hepatocyte cell line
(2) A suitable bioreactor environment and peripheral control systems
(3) An effective extracorporeal circulatory system to incorporate an artificial liver system. Though artificial support systems for kidney failure have been widely available for the past several decades, it is only recently that they have become a promising treatment modality for liver failure. The various liver support systems include conventional dialysis, charcoal hem perfusion, high volume plasma exchange, liver dialysis using sorbent technology, molecular reabsorptions recirculating system using Albumins as the dialysate, bio artificial livers, extracorporeal liver assist device and extracorporeal organ perfusion.

Keywords—Bridging, Albumins, transplantation, extracorporeal

Introduction
Unlike renal failure, artificial support systems were not widely used in liver failure, mainly because hepatic toxins are albumin-bound unlike most uremic toxins and hence cannot be removed by conventional dialysis. It has been only recently that advances have been made concerning removal of hepatic toxins. It is thus now possible to support the patient with liver failure till the liver recovers or until liver transplantation is feasible. This article describes the various methods of artificial liver support systems and highlights the recent developments in this field. The liver weight can vary from 1.2-1.6kg and it consists of four lobes of unequal size and shape that are connected to the hepatic artery and the portal vein. The hepatic artery carries oxygen rich blood from the aorta where as the portal vein carries nutrient rich blood from the small intestine. These vessels eventually divide into capillaries that are fed into thousands of lobules in the liver that are rich in hepatic cells.
Hepatic cells are essential to a healthy liver because they make up 80% of the liver volume and are responsible for all liver functions.
The liver is vital for survival and can be affected by many kinds of infections that can lead to disease that may require transplantation and/or liver therapy. The most common include; hepatitis A thru E, alcohol damage, fatty liver, cirrhosis, cancer and drug damage. If a transplant is not available, bio-artificial livers can help lengthen the survival time for patients with liver complications.

Anatomy of the liver:
The liver is located in the upper right-hand portion of the abdominal cavity, beneath the diaphragm, and on top of the stomach, right kidney, and intestines. Shaped like a cone, the liver is a dark reddish-brown organ that weighs about 3 pounds.
There are two distinct sources that supply blood to the liver, including the following: Oxygenated blood flows in from the hepatic artery, nutrient-rich blood flows in from the hepatic portal vein.
The liver holds about one pint (13 percent) of the body's blood supply at any given moment. The liver consists of two main lobes, both of which are made up of thousands of lobules. These lobules are connected to small ducts that connect with larger ducts to ultimately form the hepatic duct. The hepatic duct transports the bile produced by the liver cells to the gallbladder and duodenum.
The liver can lose three-quarters of its cells before it stops functioning. In addition, the liver is the only organ in the body that can regenerate itself.

**Functions of the liver:**

The liver regulates most chemical levels in the blood and excretes a product called bile, which helps carry away waste products from the liver. All the blood leaving the stomach and intestines passes through the liver. The liver processes this blood and breaks down the nutrients and drugs into forms that are easier to use for the rest of the body. More than 500 vital functions have been identified with the liver. Some of the more well-known functions include the following:

- Production of bile, which helps carry away waste and break down fats in the small intestine during digestion
- Production of certain proteins for blood plasma
- Production of cholesterol and special proteins to help carry fats through the body
- Conversion of excess glucose into glycogen for storage (glycogen can later be converted back to glucose for energy)
- Regulation of blood levels of amino acids, which form the building blocks of proteins
- Processing of haemoglobin for use of its iron content (the liver stores iron)
- Conversion of poisonous ammonia to urea (urea is an end product of protein metabolism and is excreted in the urine)
- Clearing the blood of drugs and other poisonous substance
- Resisting infections by producing immune factors and removing bacteria from the bloodstream
- When the liver has broken down harmful substances, its by-products are excreted into the bile or blood. Bile by-products enter the intestine and ultimately leave the body in the form of feces. Blood by-products are filtered out by the kidneys, and leave the body in the form of urine.

**Liver failure** Liver failure occurs when large parts of the liver become damaged beyond repair and the liver is no longer able to function.

- Liver failure is a life-threatening condition that demands urgent medical care. Most often, liver failure occurs gradually and over many years. However, a more rare condition known as acute liver failure occurs rapidly (in as little as 48 hours) and can be difficult to detect initially.
- The most common causes of chronic liver failure (where the liver fails over months to years) include:
  - Hepatitis B
  - Hepatitis C
- Long term alcohol consumption
- Cirrhosis
- Hemochromatosis (an inherited disorder that causes the body to absorb and store too much iron)
- Malnutrition
- The causes of acute liver failure, when the liver fails rapidly, however, are often different. These include: Acetaminophen (Tylenol) overdose.
- Viruses including hepatitis A, B, and C (especially in children). Reactions to certain prescription and herbal medications, Ingestion of poisonous wild mushroom

**Liver transplantation**

Liver transplantation or hepatic transplantation is the replacement of a diseased liver with a healthy liver from another person (allograft). The most commonly used technique is orthotopic transplantation, in which the native liver is removed and replaced by the donor organ in the same anatomic location as the original liver. Liver transplantation is a viable treatment option for end-stage liver disease and acute liver failure. Typically three surgeons and two anaesthesiologists are involved, with up to four supporting nurses. The surgical procedure is very demanding and ranges from 4 to 18 hours depending on outcome. Numerous anastomoses and sutures, and many disconnections and reconnections of abdominal and hepatic tissue, must be made for the transplant to succeed, requiring an eligible recipient and a well-calibrated live or cadaveric donor match.

### Conclusion

Bio-artificial livers can help a patient live long enough until a replacement liver becomes available. In addition, due to the
livers unique regenerative abilities, bio-artificial livers can allow a liver to heal itself by providing support in accomplishing the natural functions of the liver. Although there is not currently an artificial liver device on the market that can take the place of a liver, bio-artificial livers ultimately serve as a means of therapy for patients with liver diseases.

Bio-artificial livers in order to be successful must provide at least 10% of liver functioning, which requires approximately 10^10 hepatocytes. A major obstacle when considering patients with liver complications can only have one liter of blood and plasma drawn.

REFERENCE

- http://medicalcenter.osu.edu/patientcare/healthcare_services/liver_biliary_pancreatic_disease/liver_anatomy_function/Pages/index.aspx
- http://artificial-liver.blogspot.in/2009_11_01_archive.html
- www.biomedical.com
- www.google.com
Monitoring of Coma Patient
Govind Mahato, Kapil Anandrao
Student Of Biomedical Engineering ,Dbncoet, Yavatmal Amravati University

Abstract

Embedded system is becoming an integral part of Engineering design process for efficient analysis and effective operation. From data analysis to hardware work, everywhere embedded products are the main interest because of its reliability and time bound perfection. Coma is a prolonged period of unconsciousness. Unconsciousness is the lack of appreciation of (or reaction to) a stimulus. Coma differs from sleep in that one cannot be aroused from a coma. Coma involves two different concepts:

1.) Reactivity: Reactivity refers to the innate (or inborn) functions of the brain, i.e., the telereceptors (eyes and ears), the nociceptors (responses to pain), the arousal reaction (wakefulness) and the orienting response (turning one's head toward the source of sound or movement). We could also refer to these as reflexive movements.

2.) Perceptivity: Perceptivity refers to the responses of the nervous system to stimuli, which have been learned or acquired, i.e., language, communication skills, individual methods of movement such as gestures, etc. Perceptivity also refers to less complex learned or acquired reactions such as flinching when threatened. We can also think of these as conscious movements.

A person in a coma does not exhibit reactivity or perceptivity. He/she cannot be aroused by calling his/her name or in response to pain. As a person begins to emerge from a coma, they may begin to react to certain stimuli. To regain "consciousness" however, reactivity and perceptivity must both be present. These two elements are necessary for a state of awareness. Often, many of the elements of perceptivity must be relearned, such as speech, self-care, etc.

It becomes necessary to observe reactivity & perceptivity of coma patient 24x7 hours but manually it is impossible. The paper on “MONITORING OF COMA PATIENT” prepared by us can be used to monitor coma patient continuously.

Keywords: Telereceptor, Nociceptor, Perceptivity

Introduction

Continuous monitoring in daily life is important for the health condition control. However, portable or wearable devices need to carry by user on their own will. On the other hand, implantation sensors are not adoptable, because the generic users dislike inserting any object in the body for monitoring. Therefore, another monitoring system of the health condition is necessary. It is well known that each & every movement for a patient is very important. Coma patient monitoring by using image processing for doctors provides solution for this working in multispecialty hospitals where doctors are not able to supervise each patient every moment. Coma may occur as a complication of an underlying illness, or as a result of injuries, such as head trauma. It continuously provides following information to doctors:

1. Facial motion detection of patient
2. Heartbeats of patient
3. Body temperature of patient
4. For heartbeats measurement of patient Coma patient monitoring by image processing uses heartbeat sensor applied on finger. Body temperature of patient is measured with the help of temperature sensor. Also Coma patient monitoring by image processing uses MATLAB software for sensing the body movement as well as facial expression of the patient in hospital.

SYSTEM DESIGN AND IMPLIMENTATION

For the design and development of a system, the methodology involves with the hardware implementation. The actual implementation of the system involves in the order given below:

1) System Design: Broad definition of system hardware including sensors, microcontroller and its interface with display.
2) Circuit Design: Selection of ATMEGA8 Microcontroller and interfacing device, as per system definition. Design of hardware circuit and its testing on laboratory kits with some simple microcontroller software routines.
3) Hardware Modifications: Making any hardware changes found necessary after the initial hardware test, to produce a revised circuit board schematic diagram and layout.
4) Integration and Final Testing: Integrating the entire hardware and software modules and its final testing for data logging operations.

BLOCK DIAGRAM

Block diagram of coma patient monitoring system

Description  The camera which is placed above the patient continuously acquires the images of patient. The camera takes images according to preset values. Then these images are processed in MATLAB software in laptop or PC where
subtraction of these images is done. If there are some changes it will display motion in command window otherwise it’ll display no motion. When there is motion the signal is sent to microcontroller for blowing the buzzer via serial communication using MAX232 logic converter. It converts the CMOS/TTL logic levels to RS232 logic levels.

The heartbeat sensor counts the beats for every 30 seconds and it will multiply total count in 30 sec. by 2 and gives the output on LCD. If the beats value goes beyond the preset normal values then buzzer will ring. It is same for temperature sensor also it will measures the body temperature continuously. If the value is more than preset normal value then buzzer blow

**CIRCUIT DIAGRAM**

![Circuit Diagram]

**ADVANTAGES**

1. Low cost
2. Less power consumption
3. Portable
4. Less Complicated

Easy to handle and small size so can be move anywhere.

**APPLICATION**

1. **Facial motion detection**
   
   It detects the facial motions of comma patient with the help of image processing.

2. **Heartbeat monitoring**
   
   It continuously monitors the heartbeat of comma patient by using IR sensor.

3. **Temperature measurement**
   
   It measures the temperature of the patient.

**SYSTEM SNAPSHOTS**

![System Snapshot 1]

**WORKING SYSTEM**

![System Snapshot 2]

**SNAPSHOT**

**CONCLUSION**

From this monitoring system it can be concluded that, quick treatment to critical patient in the hospital can be provided. Thus the valuable time of the doctor is saved & also the appointment of special supervisor for each patient is also avoided.
FUTURE EXPANSIONS

Tele-monitoring system

Tele-monitoring system is mainly used for communication as well as monitoring purpose from distance place. In case of coma patient monitoring system we can use this system as tele-monitoring system by adding communication devices like CDMA or Wi-Fi, so from anywhere the physicians or doctors can monitor the patient and can take particular action.

Multipurpose system

This system can be added with multipara devices by doing some advancement, which will continuous monitors patient, stores all the data and application needing data connected directly to monitoring devices.

REFERENCES

- www.mathworks.com
- www.sunrom.com
- www.national.com
- Yan-Fang Li et al., “A low-power 20GHz static frequency divider with programmable input sensitivity”, proc. RFIC/IMS Symposium, pp. 235-238, 2003
The Study of Fluid and Mechanics

Mr. Prayas R. Patle (Department of Mechanical Engineering, 2nd year),
Ms. Sneha R. Dafale (Department of Biomedical Engineering, 2nd year),
Ms. Namita N. Dhabale (Department of Computer Engineering, 2nd year),
Dr. Bhauasaheb Nandurkar College of Engineering & Technology, Yavatmal.

ABSTRACT: Fluid mechanics is an “ancient science” that is incredibly alive today. The modern technologies require a deeper understanding of the behavior of real fluids; on the other hand new discoveries often pose new challenging mathematical problems. The aim of this thesis is to furnish some results in very different areas that are linked by the common scope of giving new insight in the field of fluid mechanics. Since the arguments treated are various, an extensive bibliography has been added. For the sake of completeness, there is an introductory chapter and each subsequent new topic is illustrated with the will of a self-contained exposition.

I. INTRODUCTION
This short script gives an introduction to fluid mechanics, the physics of moving fluids. Hydraulic and pneumatic power is widely used in the operation of engineering systems. The brakes on motor vehicles, railcar doors and hydraulic actuators and presses are typical examples. Fluid power is also widely used on aircraft, particularly for lowering and raising the undercarriage and for operating the flight control surfaces. The study of this unit will introduce learners to a range of concepts and applications of fluid mechanics that will enable them to solve engineering problems associated with fluid systems. The unit will provide learners with an understanding of surface tension and the viscous behaviour of Newtonian fluids that will then be used to determine a range of parameters in bearing systems. Learners will be introduced to the characteristics and behaviour of fluids at rest and will apply this knowledge to the inputs and outputs of hydraulic devices and systems, as well as to the determination of thrust forces and pressures that act on immersed rectangular and circular surfaces. However the field of fluids in motion applies to many things in current research. We refer the interested reader to that excellent book.

II. FLUIDS
A fluid is a substance that may flow. That is, the particles making up the fluid continuously change their positions relative to one another. Fluids do not offer any lasting resistance to the displacement of one layer over another when a shear force is applied. This means that if a fluid is at rest, then no shear forces can exist in it, which is different from solids; solids can resist shear forces while at rest. To summarize, if a shear force is applied to a fluid it will cause flow. Recall the example in class when a book was placed between my hands that were previously moving parallel to one another, even in the presence of the fluid, air. The book was somewhat distorted by the shear forces exerted on it by my hands, but eventually adopted a deformed position that resisted the force.
A further difference between solids and fluids is that a solid has a fixed shape whereas a fluid owes its shape at any particular time to that of the vessel containing it.
III. DEFINITION OF FLUID MECHANICS
Fluid mechanics is that branch of applied mechanics that is concerned with the statics and dynamics of liquids and gases. The analysis of the behavior of fluids is based upon the fundamental laws of applied mechanics that relate to the conservation of mass, energy and momentum. The subject branches out into sub-disciplines such as aerodynamics, hydraulics, geophysical fluid dynamics and bio-fluid mechanics.

IV. FLUID MECHANICS:
The study of forces that develop when an object moves through a fluid medium.
• Two fluids of interest
  – Water
  – Air
• In some cases, fluid forces have little effect on an object’s motion (e.g., shot-put)
• In other cases, fluid forces are significant – badminton, baseball, swimming, cycling, etc.
• Three major fluid forces of interest:
  – Buoyancy
  – Drag
  – Lift

V. FLUID PROPERTIES

A) Density
Density is the ratio of the mass of a given amount of the substance to the volume it occupies. Mean density is defined as the ratio of a given amount of a substance to the volume that this amount occupies. The density is said to be uniform if the mean density in all parts of the substance is the same.

B) Buoyancy
• Associated with how well a body floats or how high it sits in the fluid.
  • Archimedes’ principle: anybody in a fluid medium will experience a buoyant force equal to the weight of the volume of fluid which is displaced.
Example:
A boat on a lake. A portion of the boat is submerged and displaces a given volume of water. The weight of this displaced water equals the magnitude of the buoyant force acting on the boat.
  – The boat will float if its weight in air is less than or equal to the weight of an equal volume of water.
  • Buoyancy is closely related to the concept of density.
    Density = mass/volume

C) Drag
• Resistive force acting on a body moving through a fluid (air or water).
  Two types:
  a) Surface drag: depends mainly on smoothness of surface of the object moving through the fluid.
     • shaving the body in swimming; wearing racing suits in skiing and speed skating.
  b) Form drag: depends mainly on the cross-sectional area of the body presented to the fluid
     • Bicyclist in upright v. crouched position
     • Swimmer: related to buoyancy and how high the body sits in the water.

Fig. drag force in aeroplane
D) Lift
- Represents a net force that acts perpendicular to the direction of the relative motion of the fluid;
- Created by different pressures on opposite sides of an object due to fluid flow past the object
Example: Airplane wing (hydrofoil)

![Lift force in aeroplane](image)

Fig. lift force in aeroplane

VI. Bernoulli’s principle:
Bernoulli’s principal stated as the velocity of a flowing fluid is inversely proportional to its pressure.
- Fast relative velocity → lower pressure
- Slow relative velocity → higher pressure

![Bernoulli’s Principle](image)

Fig. Bernoulli’s principal

i. COMPRESSIBILITY
The degree of compressibility of a substance is characterized by the bulk modulus of elasticity, K, defined as:

\[ K = \frac{-\delta p}{\delta V/V} \]

Where \( \delta p \) represents the small increase in pressure applied to the substance that causes a decrease of the volume by \( \delta V \) from its original volume of \( V \). Note the negative sign in the definition to ensure that the value of \( K \) is always positive. \( K \) has the same dimensional formula as pressure, which is: \([\text{ML}^{-1}\text{T}^{-2}]\).

\( K \) can also be expressed as a function of the accompanying change in density caused by the pressure increase. Using the definition of density as mass/volume. Note that the value of \( K \) depends on the relation between pressure and density under which the compression occurs.

The reciprocal of the bulk modulus is compressibility.

ii. SURFACE TENSION
Surface tension is the surface force that develops at the interface between two immiscible liquids or between liquid and gas or at the interface between a liquid and a solid surface. Because of surface tension, small water droplets, gas bubbles and drops of mercury tend to maintain spherical shapes. The presence of surface tension and its dynamics are due to complex interactions at the molecular level along interfaces. Away from interfaces, molecules are surrounded by like molecules on all sides and so intermolecular force interactions result in a zero net force. At interfaces, molecules interact with molecules of the same fluid on only one side. The molecules at the interfaces experience a net force that puts the interface under tension.

![Surface tension in bubble](image)

Fig. surface tension in bubble
The ultimate magnitude and direction of this tension force is determined not only by what happens on either side of the interface, but by the way molecules of the two fluids interact with each other. Surface tension, therefore, is specific to the participating fluids. Surface tension forces are also sensitive to the physical and chemical condition of the solid surface in contact, such as its roughness, cleanliness, or temperature.

Dimensional Formula:
\[
\frac{[MLT^{-2}]}{[L]} = [MT^{-2}]
\]

A common symbol for surface tension is \( \sigma \).

iii. VAPOUR PRESSURE

At the surface of a liquid, molecules are leaving and re-entering the liquid mass. The activity of the molecules at the surface creates a vapour pressure, which is a measure of the rate at which the molecules leave the surface. When the vapour pressure of the liquid is equal to the partial pressure of the molecules from the liquid which are in the gas above the surface, the number of molecules leaving is equal to the number entering. At this equilibrium condition, the vapour pressure is known as the saturation pressure.

The vapour pressure depends on the temperature, because molecular activity depends upon heat content. As the temperature increases, the vapour pressure increases until boiling is reached for the particular ambient atmospheric pressure.

Dimensional Formula:
\[
[ML^{-1}T^{-2}]
\]

iv. VISCOSITY

Viscosity can be thought of as the internal “stickiness” of a fluid. It is one of the properties that control the amount of fluid that can be transported in a pipeline during a specific period of time. It accounts for the energy losses associated with the transport of fluids in ducts, channels and pipes. Further, viscosity plays an important role in the generation of turbulence. Needless to say, viscosity is an extremely important fluid property in our study of fluid flows.

All real fluids resist any force tending to cause one layer to move over another, but the resistance occurs only when the movement is taking place. From experiments with various fluids, Sir Isaac Newton postulated that for the straight and parallel motion of a given fluid, the tangential stress between two adjoining fluid layers is proportional to the velocity gradient in a direction perpendicular to the layers.

That is:
\[
\tau = \mu \frac{\partial u}{\partial y}
\]
Where $\mu$ is a constant for a particular fluid at a particular temperature. The coefficient of proportionality is the absolute viscosity (sometimes referred to as the coefficient of viscosity). Note that $\mu$ is a scalar quantity, while the other terms are vector quantities. Note also that the surface over which the stress acts is perpendicular to the velocity gradient.

v. PRESSURE

To define pressure, consider some imaginary surface of area $A$ at an arbitrary part of a fluid. This surface must experience forces, say of magnitude $F$, and due to a very large number of molecular collisions from the fluid adjoining it. Pressure, which is a scalar quantity, is defined as the ratio of the force and the area, that is $F/A$.

Dimensional Formula is:

$[ML^{-1}T^{-2}]$

CONCLUSION:

In this content we are studying the basic concept of related to the fluid power and also try to search and finding application of the concept of fluid mechanics.

REFERENCE:
1. Fundamental Concepts in Fluid Mechanics
2. Uncertainty Analysis for Fluid Mechanics with Applications
   Robert W. Waiters, Virginia Polytechnic Institute and State University, Blacksburg, Virginia, Luc Huyse
   Southwest Research Institute, San Antonio, Texas
   February 2002
ABSTRACT:-Within The Wide Range Of Issue Related To Nano Technology We Highlight Some Segment Of Nanotechnology That Will Be Based On Nano Science Nanotechnology is one of the leading scientific fields today since it combines knowledge from the fields of Physics, Chemistry, Biology, Medicine, Informatics, and Engineering. It is an emerging technological field with great potential to lead in great breakthroughs that can be applied in real life, investigate and tune the properties, responses, and functions of living and non-living matter, at sizes below 100 nm. The application and use of nanomaterials in electronic and mechanical devices, in optical and magnetic components, quantum computing, tissue engineering, and other biotechnologies, with smallest features, widths well below 100 nm, are the economically most important parts of the nanotechnology nowadays and presumably in the near future. The number of nanoproducts is rapidly growing since more and more Nano-engineered materials are reaching the global market.

Keywords: Nanometrology, Nano science, Nanoscale, Nanomaterials

I. INTRODUCTION

Nanotechnology is an exciting area of scientific development which promises “more for less”. It offers ways to create smaller, cheaper, lighter and faster devices that can do more and cleverer things, use less raw materials and consume less energy. The term nanotechnology comes from the combination of two words: the Greek numerical prefix nano referring to a billionth and the word technology. As an outcome, Nanotechnology or Nano-scaled Technology is generally considered to be at a size below 0:1 \( \mu \)m or 100 nm. Nanoscale science (or nanoscience) studies the phenomena, properties, and responses of materials at atomic, molecular, and macromolecular scales, and in general at sizes between 1 and 100 nm. In this scale, and especially below 5 nm, the properties of matter differ significantly (i.e., quantum-scale effects play an important role) from that at a larger particulate scale. Nanotechnology is then the design, the manipulation, the building, the production and application, by controlling the shape and size, the properties-responses and functionality of structures, and devices and systems of the order or less than 100 nm.

Nanotechnology is considered an emerging technology due to the possibility to advance well-established products and to create new products with totally new characteristics and functions with enormous potential in a wide range of applications. In addition to various industrial uses, great innovations are foreseen in information and communication technology, in biology and biotechnology, in medicine and medical technology, in metrology, etc. Novel materials and new-engineered surfaces allow making products that perform better. Nanomaterials with unique properties such as: nanoparticles carbon nanotubes, fullerenes, quantum dots, quantum wires, nano-fibers, and nano-composites allow completely new applications to be found. Ethical and moral concerns also need to be addressed in parallel with the new developments. Nanotechnology is an exciting area of scientific development which promises “more for less”. It offers ways to create smaller, cheaper, lighter and faster devices that...
can do more and cleverer things, use less raw materials and consume less energy.

IIORITY: THE BEGINNING OF NANO-TECHNOLOGY

The concepts that seeded nanotechnology were first discussed in 1959 by renowned physicist Richard Feynman in his talk There's Plenty of Room at the Bottom, in which he described the possibility of synthesis via direct manipulation of atoms. The term "nano-technology" was first used by Norio Taniguchi in 1974, though it was not widely known.

Inspired by Feynman's concepts, K. Eric Drexler independently used the term "nanotechnology" in his 1986 book Engines of Creation: The Coming Era of Nanotechnology, which proposed the idea of a nanoscale "assembler" which would be able to build a copy of itself and of other items of arbitrary complexity with atomic control. Also in 1986, Drexler co-founded. Thus, emergence of nanotechnology as a field in the 1980s occurred through convergence of Drexler's theoretical and public work, which developed and popularized a conceptual framework for nanotechnology, and high-visibility experimental advances that drew additional wide-scale attention to the prospects of atomic control of matter.

K. Eric Drexler

For example, the invention of the scanning tunneling microscope in 1981 provided unprecedented visualization of individual atoms and bonds, and was successfully used to manipulate individual atoms in 1989. The microscope's developers Gerd Binnig and Heinrich Rohrer at IBM Zurich Research Laboratory received a Nobel Prize in Physics in 1986. Binnig, Quate and Gerber also invented the analogous atomic force microscope that year.

Fig. Buckminsterfullerene C_{60}, also known as the buckyball, is a representative member of the carbon structures known as fullerenes. Members of the fullerene family are a major subject of research falling under the nanotechnology umbrella.

Fullerenes were discovered in 1985 by Harry Kroto, Richard Smalley, and Robert Curl, who together won the 1996 Nobel Prize in Chemistry. C_{60} was not initially described as nanotechnology; the term was used regarding subsequent work with related graphene tubes (called carbon nanotubes and sometimes called Bucky tubes) which suggested potential applications for nanoscale electronics and devices.

In the early 2000s, the field garnered increased scientific, political, and commercial attention that led to both controversy and progress. Controversies emerged regarding the definitions and potential implications of nanotechnologies, exemplified by the Royal Society's report on nanotechnology. Challenges were raised regarding the feasibility of applications envisioned by advocates of molecular nanotechnology, which culminated in a public debate between Drexler and Smalley in 2001 and 2003. [11]
Meanwhile, commercialization of products based on advancements in nanoscale technologies began emerging. These products are limited to bulk applications of nanomaterials and do not involve atomic control of matter. Some examples include the Silver Nano platform for using silver nanoparticles as an antibacterial agent, nanoparticle-based transparent sunscreens, and carbon nanotubes for stain-resistant textiles. Governments moved to promote and fund research into nanotechnology, beginning in the U.S. with the National Nanotechnology Initiative, which formalized a size-based definition of nanotechnology and established funding for research on the nanoscale.

By the mid-2000s new and serious scientific attention began to flourish. Projects emerged to produce nanotechnology roadmaps which center on atomically precise manipulation of matter and discuss existing and projected capabilities, goals, and applications.

III. WHAT IS NANOTECHNOLOGY?

Nanotechnology originates from the Greek word meaning “dwarf”. A nanometre is one billionth ($10^{-9}$) of a meter, which is tiny, only the length of ten hydrogen atoms, or about one hundred thousandth of the width of a hair! Although scientists have manipulated matter at the nanoscale for centuries, calling it physics or chemistry, it was not until a new generation of microscopes was invented in the nineteen eighties in IBM, Switzerland that the world of atoms and molecules could be visualized and managed.

DEFINITION

In simple terms, nanotechnology can be defined as ‘engineering at a very small scale’, and this term can be applied to many areas of research and development from medicine to manufacturing to computing, and even to textiles and cosmetics. It can be difficult to imagine exactly how this greater understanding of the world of atoms and molecules has and will affect the everyday objects we see around us, but some of the areas where nanotechnologies are set to make a difference are described below.

NANOTECHNOLOGY is a part of science and technology about the control of matter on the atomic and molecular scale - this means things that are about 100 nanometers or smaller. Nanotechnology includes making products that use parts this small, such as electronic devices, catalysts, and sensors etc. Nanotechnology is defined as the study of structures between 1 nanometre and 100 nanometer’s in size. To give you an idea of how small that is, there are more nanometers’ in an inch than there are inches 400 miles. To give an international idea of how small that is, there are as many nanometres in a centimetre, as there are centimetres in 100 kilometres. Nanotechnology brings together scientists and engineers from many different subjects, such as applied physics, materials science, interface and colloid science, device physics, chemistry, supra-molecular chemistry (which refers to the area of chemistry that focuses on the non-covalent bonding interactions of molecules), self-replicating machines and robotics, chemical engineering, mechanical engineering, biology, biological engineering, and electrical engineering.

Fig. Typical nanostructure geometries.
Generally, when people talk about nanotechnology, they mean structures of the size 100 nanometers or smaller. There are one million nanometers in a millimeter. Nanotechnology tries to make materials or machines of that size. People are doing many different types of work in the field of nanotechnology. At the more "science fiction" end of the field are attempts to make small copies of bigger machines or really new ideas for structures that make themselves. Nanotechnology can help in solving serious humanity problems such as energy adequacy, climate change or fatal diseases: “Nanotechnology" Alcatel-Lucent is an area which has highly promising prospects for turning fundamental research into successful innovations. Not only to boost the competitiveness of our industry but also to create new products that will make positive changes in the lives of our citizens, be it in medicine, environment, electronics or any other field. Nanosciences and nanotechnologies open up new avenues of research and lead to new, useful, and sometimes unexpected applications. The development of specific guidance documents at a global level for the safety evaluation of nanotechnology products is strongly recommended. Ethical and moral concerns also need to be addressed in parallel with the new developments. Huge aspirations are coupled to nanotechnological developments in modern medicine. The potential medical applications are predominantly in diagnostics (disease diagnosis and imaging), monitoring, the availability of more durable and better prosthetics, and new drug-delivery systems for potentially harmful drugs. While products based on nanotechnology are actually reaching the market, sufficient knowledge on the associated toxicological risks is still lacking. Reducing the size of structures to nano-level results in distinctly different properties. As well as the chemical composition, which largely dictates the intrinsic toxic properties, very small size appears to be a dominant indicator for drastic or toxic effects of particles. From a regulatory point of view, a risk management strategy is already a requirement for all medical technology applications. In order to discuss the advances of nanotechnology in nanostructured materials,

IV. FUNDAMENTAL CONCEPTS

Nanotechnology is the engineering of functional systems at the molecular scale. This covers both current work and concepts that are more advanced. In its original sense, nanotechnology refers to the projected ability to construct items from the bottom up, using techniques and tools being developed today to make complete, high performance products. One nanometer (nm) is one billionth, or $10^{-9}$, of a meter. By comparison, typical carbon-carbon bond lengths, or the spacing between these atoms in a molecule, are in the range 0.12–0.15 nm, and a DNA double-helix has a diameter around 2 nm. On the other hand, the smallest cellular life-forms, the bacteria of the genus Mycoplasma, are around 200 nm in length. By convention, nanotechnology is taken as the scale range 1 to 100 nm following the definition used by the National Nanotechnology Initiative in the US. The lower limit is set by the size of atoms (hydrogen has the smallest atoms, which are approximately a quarter of a nm diameter) since nanotechnology must build its devices from atoms and molecules. The upper limit is more or less arbitrary but is around the size that phenomena not observed in larger structures start to become apparent and can be made use of in the nano device. These new phenomena make nanotechnology distinct from devices which are merely miniaturized versions of an equivalent macroscopic device; such devices are on a larger scale and come under the description of micro technology. To put that scale in another context, the comparative size of a nanometer to a meter is the same as that of a marble to the size of the earth. Or another way of putting it: a nanometer is the amount an average man's beard grows in the time it takes him to raise the razor to his face. Two main approaches are used in nanotechnology. In the "bottom-up" approach, materials and devices are built from molecular components which assemble themselves
chemically by principles of molecular recognition.

1) V. PHYSICAL CHARACTERISTIC OF NANOMATERIAL

At Nano scale physical properties of system or particles substantially change. Physical properties such as quantum size effects where electrons move different for very small sizes of particle. Properties such as mechanical, electrical and optical changes when macroscopic system changes to microscopic one which is of utmost importance. Nano materials and particles can act as catalyst to increase the reaction rate along with that produce better yield as compared to other catalyst. Some of the most interesting properties when particle gets converted to nano scale are substances which usually stop light become transparent (copper); it becomes possible to burn some materials (aluminum); solids turn into liquids at room temperature (gold); insulators become conductors (silicon). A material such as gold, which does not react with other chemicals at normal scales, can be a powerful chemical catalyst at nanoscales. These special properties which we can only see at the nano scale are one of the most interesting things about nanotechnology.

VI. NANOMATERIALS

The nanomaterials field includes subfields which develop or study materials having unique properties arising from their nanoscale dimension. Products containing engineered nanomaterials are already in the market. The range of commercial products available today is very broad, including metals, ceramics, polymers, smart textiles, cosmetics, sunscreens, electronics, paints and varnishes. Nanomaterials must be examined for potential effects on health as a matter of precaution, and their possible environmental impacts. The development of specific guidance documents at a global level for the safety evaluation of nanotechnology products is strongly recommended.

1) Nanoscale materials can also be used for bulk applications; most present commercial applications of nanotechnology are of this flavor.
2) Progress has been made in using these materials for medical applications; see Nanomedicine.
3) Nanoscale materials such as nanopillars are sometimes used in solar cells which combats the cost of traditional Silicon solar cells.

Development of applications incorporating semiconductor nanoparticles to be used in the next generation of products, such as display technology, lighting, solar cells and biological imaging; see quantum dots.

FACTS

1) One nanometer (nm) is $10^{-9}$ or 0.000,000,001 meter.
2) When two carbon atoms join together to make a molecule the distance between them is in the range of 0.12-0.15 nm.
3) DNA double helix is about 2 nm from one side to the other. It develops into a new field of DNA nanotechnology. In future DNA can be manipulated that can lead to new revolution. Human genome can be manipulated according to requirements.
4) The bacterium is the smallest living thing with a cell is about 200 nm long.
5) A nanometer and a meter can be understood as the same size-difference as between golf ball and the Earth.
6) One nanometer is about one twenty-five-thousandth the diameter of a human hair.
7) Fingernails grow one nanometer per second.

VII. NANOTECHNOLOGY TOOLS:

1. NANOMETROLOGY
The great development in Nanotechnology has given birth to the need of knowing the dimensions that characterize its nanostructure. This lead to the appearance of a new scientific field called Nanometrology. Nanometrology is the science and practice of measurement of functionally important, mostly dimensional parameters and components with at least one critical dimension which is smaller than 100 nm. Success in nano-manufacturing of devices will rely on new nanometrologies needed to measure basic materials properties including their sensitivities to environmental conditions and their variations, to control the nanofabrication processes and materials functionalities, and to explore failure mechanisms. In order to study and explore the complex nanosystems, highly sophisticated experimental, theoretical, and modeling tools are required.

VIII. EXPONENTIAL PROLIFERATION

WHY NANOTECHNOLOGY IMPORTANT?

Nanotechnology not only will allow making many high-quality products at very low cost, but it will allow making new nanofactories at the same low cost and at the same rapid speed. This unique (outside of biology, which is) ability to reproduce its own means of production is why nanotech is said to be an exponential technology. It represents a manufacturing system that will be able to make more manufacturing systems—factories that can build factories rapidly, cheaply, and cleanly.

IX. IS NANOTECHNOLOGY BAD OR GOOD?

Nanotechnology offers great potential for benefit to humankind, and also brings severe dangers. While it is appropriate to examine carefully the risks and possible toxicity of nanoparticles and other products of nanoscale technology, the greatest hazards are posed by malicious or unwise use of molecular manufacturing. CRN's focus is on designing and promoting mechanisms for safe development and effective administration of MM.

X. LOOKING TO THE FUTURE

The total societal impact of nanotechnology is expected to be greater than the combined influences that the silicon integrated circuit, medical imaging, computer-aided engineering, and man-made polymers have had in this century. Significant improvements in performance and changes of manufacturing paradigms will lead to several industrial revolutions in the 21st century. Nanotechnology will change the nature of almost every human-made object. The major questions now are how soon will these revolutions arrive, who will benefit the most, and who will be in position to control or counter their negative aspects? How can we embrace and facilitate the new industrial revolution to maximize the benefit to US citizens? We believe that a national initiative is required to advance this goal because the needs for and from nanotechnology transcend anything that can be supplied by traditional academic disciplines, national laboratories, or even entire industries.

New medical treatments are emerging for fatal diseases, such as brain tumours and Alzheimer’s disease. Computers are built with nanoscale components and improving their performance depends upon shrinking these dimensions yet further".

XI. APPLICATION:-

Further applications allow tennis balls to last longer, golf balls to fly straighter and even bowling balls to become more durable and have a harder surface. Trousers and socks have been infused with nanotechnology so that they will last longer and keep people cool in the summer. Bandages are being infused with silver nanoparticles to heal cuts faster. Cars are being manufactured with nanomaterials so they may need fewer metals and less fuel to operate in the future. Video game consoles and personal
computers may become cheaper, faster, and contain more memory thanks to nanotechnology. Nanotechnology may have the ability to make existing medical applications cheaper and easier to use in places like the general practitioner's office and at home.

XII. REFERENCES
3. Introduction To Nanotechnology Henrik Bruus
   Mic–Department Of Micro And Nanotechnology, Technical University Of Denmark
Abstract- This paper reviews and discusses the basics and applications of RP techniques in dentistry: (1) construction of a computer aided design (CAD) model, including data acquisition, data processing, and the corresponding machines and CAD packages, (2) typical RP systems and how to choose them, and (3) current and potential use of RP techniques in dentistry. Practical application examples of RP techniques in dentistry are provided. The use of physical models provides added values in these applications. Rapid prototyping (RP) techniques have long been employed to build complex 3D models in medicine. However, publications regarding the dental application of RP technologies are still rare.

Biomedical Engineering is the application of Engineering Principles and Techniques to the Medical fields. It combines the design and problem solving skill of Engineering with Medical and Biological Sciences to improve healthcare, diagnosis and treatments. Medical imaging has been used to provide information for diagnostic and therapeutic purposes.

This paper gives an overview of the growth and trend of the technology, area of application and its significant benefits to Biomedical Engineering.

Keywords- Rapid Prototyping Technology; Computer Aided Design; 3D Models; Biomedical Engineering; Rapid Prototyping Application; Dentistry Science.

I. INTRODUCTION

In recent years, new techniques have been developing to improve the healthcare facilities. Technological development has improve the quality of operations, reduce the risk to patients and reduce the pain experienced by patients, main technique like manually invasive surgery, robot assisted operation, Computer assisted surgery and virtual reality. This has helped surgeons operate under difficult visual condition.

Rapid Prototyping (RP) Technology used in Mechanical Engineering has also found application in Medical Science. The RP Technology able to fabricate a representative, physical 3D model. The Technique aid improved visualization of anatomical features, communication between Doctor, rehearsal of complex surgical procedure and prototype for implants. The main advantage of RP is that Medical models can be created that have undercuts, voids, and complex internal geometries, such as neurovascular canals or sinuses.

The aim of the paper is to give an overview of the relatively infant technology, its diverse application, tremendous benefits to dentistry Science in Biomedical Engineering and the problems facing the technology. The paper highlight the direction technology has taken in nineties an indication of its significant contribution to the dentistry Science.

II. RAPID PROTOTYPING TECHNOLOGY OVERVIEW

Rapid prototyping (RP) is the automatic construction of physical object using additive manufacturing technology. The first technique of rapid prototyping become available in the late 1980s and were used for a much range of application and are even used to manufacture production quality parts in relatively small numbers. Some sculptors use the technology to produce complex shapes of fine arts exhibitions.

A. RAPID PROTOTYPING PROCESS

They are made up of different processes some of which are-
1. Stereo lithography (SL)
2. Selective Laser Sintering (SLS)
3. Laminated Object Manufacturing (LOM)
4. Fused Deposition Model (FDM)
5. Direct Shell Production (DSP)
6. 3D Printing.

The uses of Additive Manufacturing for Rapid Prototyping takes virtual design from Computer Aided Design (CAD) or Animation modeling software transform them into thin virtual horizontal cross section and then creates successive layer until the model is complete.

B. RAPID PROTOTYPING LAYER MANUFACTURING PROCESS

Almost all Rapid Prototyping processes, either currently available commercially or under development, are based on layered manufacturing methodology in which object as a series of horizontal cross section, each one being formed individually from the relevant raw material, and bonded to preceding layer until it is completed. The main process stages involved in fabricating parts are common to most of systems, but the mechanisms by which the individual layers are created obviously depends on the particular system.

The common process stages are shown in Fig.1. The starting point for any process is the source of the abstract geometry of the object to be built from which a data set describing that geometry must then be compiled. This data must be manipulated to generate the instruction required to control the process in the final stage of actually fabricating the components.

With additive manufacturing, the machine reads in data from CAD drawing and lays down successive layer of liquid, powder, or sheet material, and in this way builds up the model from series of cross sections. These layers, which correspond to the virtual cross section from the CAD model, are joined together or fused automatically to create the final shape. RP Process are shown in the Fig.2

The standard data interface between CAD software and the machine is the STL file format. An STL file approximates the shape of a part or assembly using triangular facets. Smaller facets produce a higher quality surface.

The word ‘Rapid’ is relative: construction of a model with contemporary method can take from several hours to several days, depending on the method used and the size and complexity of the model. Additive system for RP can typically produce in a few hours, although it can vary widely depending on the type of machine being used and the size and number of models being produce simultaneously.

Some solid freeform fabrication techniques use two materials in the course of constructing parts. The first material is the part material and the second material is the support matter. The support material is later removes by heat or dissolved away with solvent water.

Traditional injection moulding can be less expensive for manufactuering polymer product in high qualities, but additive fabrication can be faster and less expensive when producing relatively small quantities of parts. 3D printer gives designers and concept development teams the ability to produces parts and concept models using desktop size printer.

A large number of competing technologies are available in the market place. As all are additive technologies, their main difference is...
found in the way layers are built to create parts. Some are melting or softening material to produce the layers (SLS, FDM) where other are laying liquid

![Fig. 2. RP Process.]

Material thermo sets that are cured with different technologies. In the case of lamination system, thin layers are cut to shapes and joined together. RP processes with material are shown in the Table.1.

<table>
<thead>
<tr>
<th>Prototyping Technology</th>
<th>Base material</th>
</tr>
</thead>
<tbody>
<tr>
<td>Selective Laser Sintering</td>
<td>Thermoplastics, Metal Powder</td>
</tr>
<tr>
<td>Fused Deposition Modeling</td>
<td>Thermoplastics, Eutectic Metal</td>
</tr>
<tr>
<td>Stereo lithography</td>
<td>Photopolymer</td>
</tr>
<tr>
<td>Laminated Object Manufacturing</td>
<td>Paper</td>
</tr>
<tr>
<td>Electron Beam Melting</td>
<td>Titanium alloys</td>
</tr>
<tr>
<td>3D Printing</td>
<td>Various Material</td>
</tr>
</tbody>
</table>

Listed below is some medical areas in which RP systems has implemented in dentistry science, and other possible areas of applications.

Biomedical Engineering is a highly interdisciplinary, influenced by various other engineering and medical fields. Due to this diversity, it is typical for a biomedical engineer to focus on a particular subfield or group of related subfields.

Biomedical Engineering has recently emerged as its own discipline, compared to many other engineering fields. Much of the work in Biomedical Engineering consists of research and development, spanning a broad array of subfields. Prominent Biomedical Engineering application include the development of biocompatible, various diagnostic and therapeutic medical device ranging from clinical equipment to micro-implant, common imaging equipment such as MRIs and EEGs, biotechnologies such as regenerative tissue growth, and pharmaceutical drugs and biopharmaceuticals.

IV. RAPID PROTOTYPING APPLICATION IN DENTISTRY SCIENCE

1. Manufacturing of Dental Devices
RP models have been widely used in industry to fabricate components of industrial devices. Similarly, RP techniques can also be used to design, develop, and manufacture dental devices and instruments. This is simply an outgrowth of
recognized engineering applications of the technology. Recent dental devices, such as the surgical fastener, have met with uncharacteristic wide ranging acceptance in the medical community partially because RP models have enabled better feedback and assistance in production [15]. To some extent, the use of RP models can offer additional functionalities to dental devices because RP models can have very complex geometry, which is otherwise difficult to fabricate using conventional methods [14].

2. Customized implant design [11, 16–18]

Dental implants have become a common and highly successful method in replacing missing teeth. Previously, surgeries were carried out using standard-sized implant parts selected from a range provided by manufacturers. This works satisfactorily for many cases, but not all. There are always cases that are outside the standard range, between sizes, or with special requirements due to disease or genetics. The standard implant needs to be customized for the individual during surgery. A patient may be left on the operating table for hours and is at risk for surgical complications. The patient will suffer more discomfort if the implant and/or fixture do not fit well. Design of individual dental implants and fixtures provides advantages. Using RP to make implants is a potential alternative to standard implants. The combination of CT/MRI scanner or laser digitizer, rapid prototyping, and CAD packages makes it possible to manufacture customized implants and fixtures that precisely fit a patient at a reasonable cost. Implants are made to fit automatically, thus less time is wasted trying to do this during surgery. No single RP techniques dominant in dental implant application. Many of the RP technologies have been applied over a range of implant applications.

3. Orthotics [16]

RP models can be used to design orthotic devices with the specific patient’s tooth alignment. By scanning a patient’s teeth using a CT/MRI scanner or laser digitizer, a model of the patient’s tooth condition can be precisely built. The specific tooth alignment characteristics for an individual are included in the prototype, allowing for development of a biomechanically correct geometry that improves the fit, comfort, and stability. Through this process, the number of times that orthotics have to be refitted is decreased, thus cutting down the overall cost.

4. Prosthetics [19–21]

Until now, dental prosthesis (coping, crown, bridge, fixture, etc.) fabrication has greatly depended on the skills of dentists and technicians. Fortuitously, RP techniques are becoming an alternative way. For example, patterns for dental crowns and implant structures can be fabricated using an RP machine. Dental crowns can be used to restore damaged or missing teeth. A dental crown model can be constructed from the inner surface to the outer. The inner geometrical data can be obtained either by scanning the surface of the tooth after tooth preparation or based on the profile of the implant, while the outer surface can be designed based on the scanned data from neighbouring teeth and the teeth on the opposite side of the mouth and aesthetics consideration. After the construction of the crown model, the model can be sliced and transferred to an is then investment cast to a metallic or ceramic crown. Through this process, the feedback on the design of the dental crown from the patient can be taken into consideration before the dental crown is fabricated.

5. Anthropology [16]

RP models can be very beneficial to the anthropologist. RP allows for replication of jawbone and teeth so that moulding, measuring, and dissecting of the remains can be done without causing harm. In cases where only one or two specimens exist, the RP models allow the original model to be safely locked away without hindering research done on the specimen. The models that are built can also be used to show changes in evolution.
6. Forensics [16, 22]

It is extremely important to keep or reconstruct the scene to investigate a crime. RP is a valuable tool in criminal investigation. RP models can be kept as evidence in criminal investigation and help investigators find more clues. RP models can be used to preserve evidence before it further deteriorates. They are accurate enough to see the effects of wounds. Furthermore, accurate predictions of the forces, implements and other key events can be determined using these models. RP allows production of simulated bones, so actual physical models are something tangible that could be presented in a court proceeding. These models could help recreate the scene in the courtroom as well as shine some light on what really happened.

7. Biologically active implants [12, 13, 23]

The manufacture of biologically active implants is a brand new area of RP application. Many investigations have been performed recently, but it will take years for this technique to be widely employed in practice [12, 13]. Similarly, in dentistry RP techniques have the potential to manufacture biologically active implants such as jawbones, which might be damaged or otherwise malformed due to disease.

VI. CONCLUSION

The following advantages can be derived from the use of RP system in biomedical engineering;

1. Awareness: Allowing a surgeon, from the outset, to know what to expect when a certain surgical route is adopted. Thus, it reduces the duration of the procedure and greatly reduces the risk of infection and the problems of uncertainly.

2. Accuracy: A mould part for artificial limbs can be CT

3. Planning: Allowing surgical dry as well as marking out vital areas to be avoided and predicting complication that in the operation, and reduces the search time of the correct entry of surgical tools.

4. Validation: A prototype can be used to match the physical dimension of the patient and then, using the measurements, predict the surgical outcome and assess post-operative changes that might arise after surgery.

5. Copy: Multiple copies of the prototype can be made for testing and planning before the actual performance of the operation.

6. Messenger: The prototype can also be used to communicate between surgeons, nurses and even immediate family member of the patient. Thus, the surgeon can better explain the procedure through the RP model than by 2D X-ray slides.

7. Timed and cost: The operation time can be greatly reduced hence reducing the use of blood transfusion.

RP techniques have been substantially employed in medicine; however, the applications of RP in dental area are relatively rare.

This paper has discussed rapid prototyping techniques and their applications in dentistry. Every RP technique starts with a CAD model. The construction of a CAD model and its associated techniques are discussed, including the data scanning modalities, data acquisition, data processing, and use of CAD packages. Both general purpose and specific software packages commercially available greatly benefit the design of CAD models. With the help of various RP techniques, the fabrication of dental objects can
be done easily and rapidly. It is otherwise difficult to generate them by NC machining because of features like overhangs, sharp corners, and undercuts in most dental objects. The diversity of RP systems and corresponding build materials makes it possible to generate dental objects with different needs (colour, transparency, and material) for different applications like custom implant design. After presentation of RP techniques, the current and potential use of RP techniques in dental application are reviewed and discussed. It is clear that the use of RP models in dental application will be expanded in the future with the ongoing development of RP techniques. A number of application examples have also been reviewed and discussed, which demonstrate that RP techniques are playing a more and more important role in dental application. RP models will benefit dentistry in many areas including surgical planning, prosthesis design, and so on.

REFERENCES


http://home.att.net/~castleisland/med_lks.htm
Bone Assessment Using Vibration Technique

Mr. Santanaka K. Karemore¹, Ms. Ankita M. Bhoot², Ms. Darshana M. Jibhkate³, ⁴ Prof Sandip M. Ghuge
Department Of Biomedical Engineering
SGBAU, Amravati (MH), India

ABSTRACT
Bone Mineral Density (BMD) is the amount of mineral per square centimetre of bone. Osteoporosis is defined as a systemic skeletal disease characterized by low bone mass and micro architectural deterioration of bone tissue, with a consequent increase in bone fragility and susceptibility to fracture. The prime motive of our work is to produce a cost effective medical instrument of the Bone Mineral Density (BMD). The developed instrumentation for assessing bone quality utilizes the principle of impulse response method. Device consists of an automated hammer which knocks in the medial side of the proximal tibia. The stress wave is propagated through the bone and it is picked up by two triaxial MEMS accelerometers. Using this setup the acceleration magnitude has been measured for various subjects and the corresponding signals were obtained. The acquired signals are fed into the computer.

Key words: MEMS, BMD, Impulse response.

I. INTRODUCTION
BMD measurements in conjunction with information about the structure and elastic properties of bone will result in a good indication of its mechanical condition and susceptibility to fracture. Moreover, during accidental impact, our bones are subjected to high strain rate loading. Since bone is a viscoelastic material, its response to this type of loading cannot be assumed to be the same as predicted by a static analysis. Therefore, it is important to study dynamic characteristics of bone under normal and diseased state in order to understand its response to more realistic loading condition. The change in Peak acceleration magnitude with age is also evaluated. Osteoporosis is a disease of the bones. It happens when you lose too much bone, make too little bone or both. As a result, bones become weak and can break from a minor fall or in serious cases, even from simple actions, like sneezing or bumping into furniture. The risk of osteoporosis fractures can be reduced with lifestyle changes and in those with previous osteoporosis related fractures medications. Lifestyle change includes diet, exercise, and preventing falls. The utility of calcium and vitamin D is questionable in most. Bisphosphonates are useful in those with previous fractures from osteoporosis but are of minimal benefit in those who have osteoporosis but no previous fractures. The diagnosis of osteoporosis can be made using conventional radiography and by measuring the bone mineral density (BMD). The most popular method of measuring BMD is dual-energy x-ray absorptiometry (DEXA). Bone quality is a composite of properties that make bone resist fracture such as its micro architecture accumulated microscopic damage the quality of collagen mineral crystal size and bone turnover. The most prevalent sequence is compression fractures of the vertebral bodies and fractures of the ribs, proximal femur (hip), humerus and distal radius. These fractures lead to deformity, loss of mobility, independence and even death. With increasing population of elderly women, the assessment and treatment of osteoporosis has become an important problem in clinical gynaecology. Bone mineral loss occurs with aging, menopause, and disuse. The decrease in biomechanical strength of bone with age is much more pronounced than the loss of bone mass due to perforations during the remodelling process. Therefore, it is important to study dynamic characteristics of bone under normal and diseased state in order to understand its response to more realistic loading condition. The change in acceleration magnitude with age is also evaluated.

II. SYSTEM DESCRIPTION
In the following subsections, we will describe the system architecture, hardware and software components,

A. SYSTEM ARCHITECTURE
As shown in the figure 1, the architecture of the system mainly consists of a mechanical section for producing knock and a electrical section for automatically operate the hammer to make knock on bone with predefined uniform force and time duration and a Data Acquisition card for interfacing the hardware part with PC. The mechanical section consists of a less weight hammer which is driven by the motor. Two actuators the accelerometer-I is placed on the fascies medialis tibia at a distance of 6cm from the medical condyle and accelerometer-II placed at a distance of 18cm from the accelerometer-I. The automated hammer that runs by the DC motor hits the Medialis Condolosis region in the leg. The automated hammer is controlled by switches. When the hammer hits the Medialis Condolosis region, the impulse response that propagates through the bone of the leg. The accelerometers will receive the impulse responses that are acquired from the hammer and are collected by the DAQ (Data Acquisition card) system. Tracer DAQ software included for acquiring and displaying data and generating signals. From the signals, we can analyse the mechanical strength of the bone.

B. HARDWARE COMPONENTS
The Hardware part consists of a microcontroller AT89552 is used for controlling the knock as desired. The microprocessor is running under the core of 8051 principle. Flash programming which is used, so that it can be made rewritable in order to increase or decrease the speed of the motor setup. It has 4 ports namely PORT 0, PORT 1, PORT 2 and PORT 3. PORT 0 is used for controlling the switches. PORT 1 is used for controlling the Digital to Analog converter. Microprocessor circuit consists of crystal oscillator used for timing control purpose. The DAC 0808 is used which has 8 data output. From the microcontroller, PORT 1 is connected to the DAC0808 combined with an Integrated circuit(IC) operational amplifier 74, for the amplification purpose of the...
Fig. 1 Schematic block diagram of the hardware setup

signals generated. DC motors are part of the electrical section using DC power as energy source. The ADXL335 is a small, thin, low power, complete 3-axis accelerometer with signal conditioned voltage outputs. The product measures acceleration with a minimum full-scale range of ±3g. It can measure the dynamic acceleration resulting from motion, shock, or vibration. The user selects the bandwidth of the accelerometer using the $C_X$, $C_Y$, and $C_Z$ capacitors at the $X_{OUT}$, $Y_{OUT}$, and $Z_{OUT}$ pins. Bandwidths can be selected to suit the application, with a range of 0.5 Hz to 1600 Hz for X and Y axes, and a range of 0.5 Hz to 550 Hz for the Z axis.

The ADXL335 is available in a small, low profile, 4 mm × 4 mm × 1.45 mm, 16-lead, plastic lead frame chip scale package (LFCSP-LQ). MEMS accelerometers and its subsystems that reliably and accurately detect and measure acceleration, tilt, shock and vibration in performance-driven applications. A fixed power of 5V dc is applied the sensor for its operation.

C. SOFTWARE COMPONENTS
USB-1208 Series DAQ modules are low-cost, PC-based analog and digital I/O devices available in USB high speed (USB-1208HS Series), full-speed (USB-1208FS/1408FS), and low-speed (USB-1208LS) models. All of these modules offer up to four DIFF or eight SE analog inputs, up to 16 digital I/O channels, and up to two counter inputs. The USB-1208LS offers two, 10-bit analog output channels with DAC rates up to 100 S/s. The USB-1208FS/1408FS both offer two, 12-bit analog output channels with DAC rates up to 10 kS/s. The USB-1208HS-2AO offers two 12-bit analog outputs and the USB-1208HS-4AO offers four, 12-bit analog outputs, each with DAC rates up to 1 MS/s. InstaCal Installation, calibration, and test software in one package simplifies these important steps as you turn your PC into a measurement system. Installation detects new hardware and configures your computer and board. Calibration software automates this critical step and keeps your measurements accurate. Test routines verify that all the board's features are operating, and will speed you to a quick resolution. Tracer DAQ is an out-of-the-box application that allows data to be generated, acquired, analyzed, displayed and exported within seconds of installing data acquisition hardware. It offers four different data acquisition applications; a Strip Chart, an Oscilloscope, a Function Generator and a Rate Generator, all of which are accessed via a common, easy-to-use menu page.

III. RESULTS
In the study, 25 subjects had participated with different sex and age group between 18-20. Before start the experimental procedure, the subjects were asked to answer the questionnaire that contains the questions regarding the age, weight height, medication, Medical history,
occupation, exercising habit etc. From that we can screen the subjects from normal and abnormal. The sampling time and sampling rates are made constant for all the subjects during acquisition of signals. We used 2 channels of the DAQ card to acquire signals from 2 sensors in Y direction. From the results obtained, we can concluded that the acceleration value obtained in terms of milli volts in channel 2 (Y-Axis of second accelerometer) is found low magnitude compared to the output of channel -1 (Y-axis of First accelerometer).

IV. CONCLUSION
A Instrumentation system is developed for assessing the bone quality and described how this can be used to diagnose and distinguish different subjects. By incorporating the precision accelerometer sensor and the automating knocking mechanism in the system good response is obtained. The experimental results have shown the usefulness of the system in assessing the quality of bone. Moreover further analysis can be done using this setup by selecting more parameters in order to strengthen the results. The technique gives a better understanding of the dynamic behaviour of bone under impact force. The study is non-invasive, reliable, easy to operate, inexpensive and has diagnostic potential in the assessment of bone quality. Further analysis has to be done by obtaining more number of data from older age group men and women (with menopause stage) and also with abnormal cases as a extension work in order to justify strongly that our system is highly reliable and can be used as a diagnosing tool for assessment of bone quality non-invasively it is required too.

REFERENCES
1. Mrs.s.mythili,development and implementation of instrumentation system for diagnosing bone quality using vibration technique
6. Evaluation and Assessment of Osteoporosis by Quantitative Ultrasound (Kaufman and Einhorn, 1993
Performance Comparison of Booth Multiplier and Wallace Tree Multiplier

Rupesh N. Gurle¹, Sanjivani P. Edakhe², Prof. Abhay R. Kasetwar³

¹B.E. final year, EXTC, DBNCOET, Yavatmal, Maharashtra, India – 445001.
²B.E. final year, EXTC, DBNCOET, Yavatmal, Maharashtra, India – 445001.
³Assistant Professor, EXTC, DBNCOET, Yavatmal, Maharashtra, India – 445001

Abstract- This paper proposes comparative analysis of booth and Wallace tree multiplier. The multiplication operation is present in many parts of digital system or digital computer, most notably in signal processing, graphics and scientific computation. With advances in technology, various techniques have been proposed to design multipliers, which offer high speed, low power consumption and lesser area. Thus making them suitable for various high speed, low power compact VLSI implementations. For arithmetic multiplication various multiplication architectures like array multiplier, Booth multiplier, Wallace tree multiplier and Booth Wallace multiplier have been thoroughly discussed. It has been found that Booth multiplier is most efficient among all, giving optimum delay, power and area for multiplication.

Keywords: Booth multiplier (radix-2), Booth multiplier (radix-4), Wallace tree multiplier.

I. INTRODUCTION

Multiplication is one of the basic functions used in digital signal processing. It requires more hardware resources and processing time than addition and subtraction. In fact, 8.72% of all instruction in atypical processing unit is multiplier [1]. Most high performance digital signal processing systems rely on hardware multiplication to achieve high data throughput. Multiplication is an important fundamental arithmetic operation. Multiplication based operations such as Multiply and Accumulate (MAC) are currently implemented in many digital signal processing applications such as convolution, Fast Fourier Transform (FFT), Filtering and in microprocessor in its arithmetic and logic unit [2]. Since multiplication dominates the execution time of most DSP algorithms, so there is need of high speed multiplier. Currently, the multiplication time is still dominant factor in determining the instruction cycle time of a DSP chip. The amount of circuitry involved is directly proportional to square of its resolution i.e., a multiplier of size of n bits has O (n²) gates. In the past, many novel ideas for multipliers have been proposed to achieve high performance. The demand for high speed processing has been increasing as a result of expanding computer operation and signal processing applications. Multiplication is a mathematical operation that is an abbreviated process of adding an integer to itself a specified number of times. A number (multiplicand) is added to itself a number of times as specified by another number (multiplier) to form a result (product). The multiplicand is multiplied by each digit of the multiplier beginning with the rightmost, LSD. Intermediate results (partial products) are placed one atop the other, offset by one digits of the same weight. The final product is determined by summation of all the partial products. Multiplication involves three main steps [3].

- Partial product generation
- Partial product reduction
- Final addition

For the multiplication of an n-bit multiplicand with an m-bit multiplier, m partial products are generated and product formed is n+m bits along.

II. BOOTH MULTIPLIER

Andrew D. Booth proposed the Booth recoding, or Booth algorithm in 1951. This
method can be used to multiply two 2’s complement number without the sign bit extension. Booth observed that when strings of ‘1’ bits occur in the multiplicand the number of partial products can be reduced by subtraction. Table 1.1 shows the Booth algorithm operation.

<table>
<thead>
<tr>
<th>$X_i$</th>
<th>$X_{i-1}$</th>
<th>Operation</th>
<th>Comments</th>
<th>$Y_i$</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
<td>Shift only</td>
<td>String of zeros</td>
<td>0</td>
</tr>
<tr>
<td>1</td>
<td>0</td>
<td>Sub and shift</td>
<td>Beg of string of ones</td>
<td>1</td>
</tr>
<tr>
<td>1</td>
<td>1</td>
<td>Shift only</td>
<td>String of zeros</td>
<td>0</td>
</tr>
<tr>
<td>0</td>
<td>1</td>
<td>Add and shift</td>
<td>End of string of ones</td>
<td>1</td>
</tr>
</tbody>
</table>

Booth classifieds group of bits into beginning, middle or end of run. String of zeros avoid arithmetic, so these can be left alone. Booth algorithm changed the original algorithm by looking at two bits of multiplier Conventional array multiplier, like the Braun multiplier and Baugh Woolley multiplier achieve comparatively good performance but they require large area of silicon, unlike the add-shift algorithms, which requires less hardware and exhibit poorer performance. The booth multipliers makes use of booth encoding algorithms in order to reduce the number of partial products by considering two bits of the multiplier at a time, thereby achieving a speed advantage over other multiplier architectures. This algorithm is valid for both signed and unsigned numbers. It accepts the number in 2’s complement form, based on radix-2 computation.

Booth multiplier reduces the number of iteration step to perform multiplication as compared to conventional step. Booth algorithm ‘scans’ the multiplier operand and skip chain of this algorithm. Booth algorithm can reduce the number of additions require to produce the result compared to conventional multiplication algorithms.

A. Booth Multiplier (Radix-2)

The Booth algorithm was invented by A. D. Booth forms the base of Signed number multiplication algorithms that are simple to implement at the hardware level, and that have the potential to speed up signed multiplication considerably. Booth’s algorithm is based upon recoding the multiplier, $y$, to a recoded, value, $z$, leaving the multiplicand, $x$, unchanged. In Booth recoding, each digit of the multiplier can assume negative as well as positive and zero values. There is a special notation, called signed digit (SD) encoding, to express these signed digits. In SD encoding +1 and 0 are expressed as 1 and 0, but -1 is expressed as 1.

The value of a 2s complement integer was defined a by equation 1.

$$y = -y_{m-1}2^{m-1} + \sum_{i=0}^{m-2}y_i2^i \quad (1)$$

This equation says that in order to get the value of a signed 2’s complement number, multiply the m-th digit by $-2^{m-1}$, and multiply each remaining digit $i$ by $+2^i$. For example, -7, which are 1001 in 2’s complement notation, would be, in SD notation, $1001 = -8 + 0 + 0 + 1 = -7$.

For implementing booth algorithm most important step is booth recoding. By booth recoding we can replace string of Is by Os. For example the value of strings of five 1s, $11111 = 2^5 - 1 = 100001 = 32 - 1 = 31$.

Hence if this number were to be used as the multiplier in a multiplication, we could replace five additions by one addition and one
subtraction. The Booth recoding procedure, then, is as follows:
1. Working from LSB to MSB, replace each 0 digit of the original number with a 0 in the recoded number until a 1 is encountered.
2. When a 1 is encountered, insert a 1 at that position in the recoded number, and skip over any succeeding 1's until a 0 is encountered.
3. Replace that 0 with a 1 and continue. This algorithm is expressed in tabular form in Table 1.

<table>
<thead>
<tr>
<th>$y_i$</th>
<th>$y_{i-1}$</th>
<th>$z_{i+1}$</th>
<th>Multiplier value</th>
<th>Situation</th>
</tr>
</thead>
<tbody>
<tr>
<td>0</td>
<td>0</td>
<td>0</td>
<td>0</td>
<td>String of 0s</td>
</tr>
<tr>
<td>0</td>
<td>1</td>
<td>1</td>
<td>+1</td>
<td>End of string of 1s</td>
</tr>
<tr>
<td>1</td>
<td>0</td>
<td>1</td>
<td>-1</td>
<td>Begin string of 1s</td>
</tr>
<tr>
<td>1</td>
<td>1</td>
<td>0</td>
<td>0</td>
<td>String of 1s</td>
</tr>
</tbody>
</table>

1) Radix-2 Booth Algorithm
1. Add zero to the right of LSB multiplier to make pairing of two bits from right to left and mark corresponding multiplier,
2. 00 or 11: do nothing.
3. 01: marks the end of a string of 1s and add multiplicand to partial product.
4. 10: marks the beginning of the string of 1s subtract multiplicand from partial product.

One of the solutions for realizing high speed multipliers is to enhance parallelism which helps in decreasing the number of subsequent calculation stages. The original version of Booth multiplier (Radix-2) had two drawbacks [4].

1. The number of add/subtracts operations become variable and hence became inconvenient while designing parallel multipliers.
2. The algorithm become inefficient when there are isolated 1s.

B. Booth Multiplier (Radix-4)

It is possible to reduce the number of partial products by half by using the technique of radix 4 booth recoding. The basic idea is that, instead of shifting and adding for every column of the multiplier term and multiplying by 1 or 0, we only take every second column, and multiply by $\pm 1, \pm 2$ or 0 to obtain the same result. Radix-4 booth encoder performs the process of encoding the multiplicand based on multiplier bits. It will compare 3 bits at a time with overlapping technique. Grouping starts from the LSB, and the first block only uses two bits of the multiplier and assumes a zero for the third bit as shown in fig.2

![Fig 2. 2 bit pairing as per booth recoding](image)

The functional operation of radix-4 booth encoder is shown in table 3. It consists of eight different types of states and during these states we can obtain the outcomes, which are multiplication of multiplicand with 0, -1 and -2 consequently.

1) Radix-4 Booth Algorithm
1. Extend the sign bit 1 position necessary to ensure that n is even.
2. Append 0 to the right of the LSB of the multiplier.
3. According to the value of each vector, each partial product will be 0, +y, -y or +2y, -2y.

The negative values of y are made by taking the 2s complement and Carry Look Ahead (CLA), fast adders are used for addition. The multiplication of y is done by shifting y by one bit to the left. Thus, in any case in designing n-bit parallel multipliers only n/2 partial products
are generated. The advantage of this method is halving of the number of partial products.

### TABLE 3
**BOOTH RECODING TABLE FOR RADIX-4**

| Multiplier bits block | Recoded 1-bit pair | 2 bit booth
<table>
<thead>
<tr>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>i+1</td>
<td>i</td>
<td>i-1</td>
</tr>
<tr>
<td>0</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>0</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>0</td>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>0</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>1</td>
<td>0</td>
<td>0</td>
</tr>
<tr>
<td>1</td>
<td>0</td>
<td>1</td>
</tr>
<tr>
<td>1</td>
<td>1</td>
<td>0</td>
</tr>
<tr>
<td>1</td>
<td>1</td>
<td>1</td>
</tr>
</tbody>
</table>

This is important in circuit design relates to the propagation delay in the running of the circuit, and the complexity and power consumption of its implementation [4].

### III. WALLACE TREE MULTIPLIER

A fast process for multiplication of two numbers was developed by Wallace. In 1964 C.S. Wallace observed that it is possible to find structure, which performs the addition operation in parallel, resulting in less delay. Wallace introduced a different way of parallel addition of the partial product bits using a tree of carry save adder, which is known as Wallace tree. A Wallace tree is an efficient hardware multiplication of a digital circuit that multiplies two integers in order to perform the multiplication of two numbers with the Wallace method. Partial product matrix is reduced to a two row matrix by using a carry save adder and the remaining two rows are summed using a fast carry propagate adder to form the product. In Wallace tree architecture, all the bits of all partial products in each column are added together by a set of counters in parallel without propagating any carries. Another set of counters then reduces this new matrix and so on, until two row matrixes are generated. Wallace method uses three steps to process the multiplication operation [5].

1. Formation of bit products.
2. The bit product matrix is reduced to a two row matrix by using a carry save adder.
3. The remaining two rows are summed using a fast carry propagate adder to produce the product.

### IV. COMPARISON OF MULTIPLIERS

Generally it is not possible to say that an exact multiplier yields to greater cost effectiveness, since trade-off is design and technology dependent. For operands of sixteen bits and higher, the modified booth algorithm or booth algorithm radix-4 reduces the partial product by half thus speed of the multiplier increases [7]. Wallace strategy for carry save adder trees is to combine the partial product bits as early as possible. This method yields to simpler CSA tree and wider carry propagate adder and the design using the Wallace tree method are fast. However a logarithmic depth reduction tree based CSA has an irregular structure that makes the design and layout difficult. Moreover connections and signal path of varying length may lead to signal skew that have implications for both power and performance. Wallace multiplier has limitation of being very irregular, so efficient layout not possible [5]. Routing between the levels become complicated. Wallace tree architecture requires long wires that having very high capacitance, comparison of booth and Wallace tree multiplier is shown in table 4.

### TABLE 4
**COMPARISON TABLE**

<table>
<thead>
<tr>
<th>Multiplier type</th>
<th>Speed</th>
<th>Circuit complexity</th>
<th>Layout</th>
<th>Area</th>
</tr>
</thead>
<tbody>
<tr>
<td>Wallace tree multiplier</td>
<td>High</td>
<td>Complex</td>
<td>More irregular</td>
<td>Large</td>
</tr>
<tr>
<td>Booth multiplier</td>
<td>Higher</td>
<td>Medium</td>
<td>Irregular</td>
<td>Smallest</td>
</tr>
</tbody>
</table>
V. CONCLUSION

This paper attempts to give a clear concept of booth multipliers and Wallace tree multiplier and their comparison. We found that the booth multipliers are better than the Wallace tree multiplier. We concluded this from the comparison based on different parameters such as speed, circuit complexity, layout and the total area. In case of booth multipliers, the total area is much less than that of Wallace tree multipliers. Hence the power consumption is also less. So we always need to find a better solution in case of multipliers. Preferably multipliers should always consume less power and cover less area. So through this paper we try to determine which of the multiplier works the best. In the end we determine that booth multiplier works the best.

VI. REFERENCES

"The Genechip System Supports both DNA and RNA Analysis"

Priti B. Walthare\!*\!1, Pratikshya R. Gahane\!*\!2*, Shilpa H. Ambule\!*\!3

\* Student Department of Biomedical Engineering, Amravati university, Yavatmal, Maharashtra, India
\# Student Department of Biomedical Engineering, Amravati university, Yavatmal, Maharashtra, India

ABSTRACT

Bioinformatics is an emerging scientific discipline representing the combined power of Biology, Mathematics, IT, Computer Science and Electronics to solve complex problems in life sciences and particularly in biotechnology. The GeneChip System supports both DNA and RNA analysis. These chip are based on the electrical and electron charges exhibited by the cells of the human body. In this paper it aimed to show the relevance of electronics in a field of bioinformatics, this paper deals with the analysis of cancer using GeneChip (DNA microarrays). Gene chip are used for Global understanding of abnormal gene expression contributing to malignancy, prediction of drug side effects during preclinical development and toxicology studies, which predicts treatment success or failure.

Keyword: GeneChip, DNA, RNA, prediction drug.

Introduction

Bioinformatics is an integrated multidisciplinary field. It comprises molecular biology (biochemistry, genetics, structural biology etc.), computer science (computational theory, artificial intelligence, machine learning, dynamic programming etc.), physical chemistry (thermodynamics, molecular modeling etc) and mathematics (algorithms, modeling, probabilistic, statistics etc.). It is hard to draw a clear line between the ‘exact disciplines’ of bioinformatics research because this is a broad and fast growing field.

A new technology, called DNA microarray, has attracted tremendous interests among biologists. This technology promises to monitor the whole genome on a single chip so that researchers can have a better picture of the interactions among thousands of genes simultaneously. Base pairing (i.e., A-T and G-C for DNA; A-U and G-C for RNA) or hybridization is the underlining principle of DNA microarray. An array is an orderly arrangement of samples. It provides a medium for matching known and unknown DNA samples based on base-pairing rules and automating the process of identifying the unknowns. An array experiment can make use of common assay systems such as microplates or standard blotting membranes, and can be created by hand or make use of robotics to deposit the sample. The sample spot sizes in microarray are typically less than 200 microns in diameter and these arrays usually contain thousands of spots. Microarrays require specialized robotics and imaging equipment. DNA microarray, or DNA chips are fabricated by high-speed robotics, generally on glass but sometimes on nylon substrates. There are two major application forms for the DNA microarray technology:

1) Identification of sequence (gene / gene mutation).
2) Determination of expression level (abundance) of genes

1. DNA Chip

Definition of DNA chip:

"A high density array of short DNA molecules bound to a solid surface for use in probing a biological sample to determine gene expression, marker pattern or nucleotide sequence of DNA/RNA” . DNA chip is also called as “Gene Chip” or “DNA microarray”. Initially developed to enhance genomic sequencing projects, especially the Human Genome Project, DNA chips are finding applications throughout the field of molecular biology. Gene scanning techniques that are based on oligonucleotide arrays called DNA chips, provide a rapid method to analyze thousands of genes simultaneously. DNA chips are thus potentially very powerful tools for
gaining insight into the complexities of gene expression, detecting genetic variations, making new gene discoveries, fingerprinting and developing new diagnostic tools. The production of DNA chips have evolved along two major pathways:

One method uses nucleic acids that have been immobilized on the chip surface sequentially to form oligonucleotides and the other method involves complementary DNA from an individual with a known genetic mutation as a source of prefabricated oligonucleotides. In either case, the problem lies with how to attach the nucleic acids or cDNA to the chip. Chips using nucleic acids are produced using photolithography. Photolithography, according to the Science article by Stephen Fodor, consists of the modification of synthetic linkers, containing photochemically removable protecting groups, attached to a glass substrate, usually a silicon-derivative glass chip. Light is directed at the photolithographic “mask” at specific areas of the chip in order to facilitate the removal of the photoactive groups, yielding 5 hydroxy groups. These modified groups are now capable of binding other nucleotides, generating a highly specific probe, which contains the sequence of a known disease causing genetic mutation. The other method, described in the DNA Chips and Microarrays website, uses purified single-stranded cDNA from an individual with a known genetic disease, requiring the use of touch or fine micropipetting, to spot the cDNA onto the surface of the chip. The cDNA immobilizes on the chip through covalent bonds, due to the positively charged surface, produced by amino silane or polylsine. For both types of chips, a potential DNA target sequence, from an asymptomatic individual, is fluorescently tagged and allowed to interact with the probes. Hybridization will occur at complementary sequences between the two samples resulting in a fluorescent image, which is then scanned by a laser beam and analyzed by a computer. The intensity of fluorescent light varies with the strength of the hybridization, thus providing a quantitative ‘snapshot’ of gene expression. This approach, requiring only minute consumption of chemical reagents and minute preparations of biological samples, can scan more than 400,000 probes placed on a single chip measuring 1.28cm X 1.28cm in size. As of now, specific chips are available for as little as $100, but could cost over thousands of dollars, once custom-made chips are available. In the future, attempts to design chips using the computer, instead of doing it by hand, will greatly speed up the process allowing companies to make custom chips in one day, as opposed to months, which would lower the cost of production. Consequently, DNA chips could probably sell for about $50, providing access to scientists.

Figure: Schematic representation of a DNA microchip

2. DNA Microarray

A DNA microarray (gene chip / DNA chip) is a collection of microscopic DNA spots attached to a solid surface. Scientists use DNA microarrays to measure the expression levels of large numbers of genes simultaneously or to genotype multiple regions of a genome. Each DNA spot contains picomoles (10^{-12} moles) of a specific DNA sequence, known as probes (or reporters). These can be a short section of a gene or other DNA element that are used to hybridize a cDNA or cRNA sample (called target) under high-stringency conditions. Probe-target hybridization is usually detected and quantified by detection of fluorophore, silver-, or chemiluminescence-labeled targets to determine relative abundance of nucleic acid sequences in the target. Since an array can contain tens of thousands of probes, a microarray experiment can accomplish many genetic test in parallel. Therefore arrays have dramatically accelerated many types of investigation.

DNA microarray can be used to measure changes in expression levels, to detect single nucleotide polymorphism (SNPs), or to genotype or resequence mutant genomes. Microarrays also differ in fabrication, workings, accuracy, efficiency, and cost. Additional factors for microarray experiments are the experimental design and the methods of analyzing the data.

2.1 What Exactly Is a DNA Microarray

DNA Microarrays are small, solid supports onto which the sequences from thousands of different genes are immobilized, or attached, at fixed locations. The supports themselves are usually glass microscope
slides, the size of two side-by-side pinky fingers, but can also be silicon chips or nylon membranes. The DNA is printed, spotted, or actually synthesized directly onto the support. With the aid of a computer, the amount of mRNA bound to the spots on the microarray is precisely measured, generating a profile of gene expression in the cell.

The American Heritage Dictionary defines "array" as "to place in an orderly arrangement". It is important that the gene sequences in a microarray are attached to their support in an orderly or fixed way, because a researcher uses the location of each spot in the array to identify a particular gene sequence. The spots themselves can be DNA, cDNA, or oligonucleotides. There are two major application forms for the DNA microarray technology:

1) Identification of sequence (gene / gene mutation)

2) Determination of expression level (abundance) of genes of one sample or comparing gene transcription in two or more different kinds of cells.

2.2 Why Are Microarrays Important?

What are Microarrays will be described in the next paragraph. Before the actual definition, it is important to understand that microarrays are a significant advance both because they may contain a very large number of genes and because of their small size. Microarrays are therefore useful when one wants to survey a large number of genes quickly or when the sample to be studied is small. Microarrays may be used to assay gene expression within a single sample or to compare gene expression in two different cell types or tissue samples, such as in healthy and diseased tissue. Because a microarray can be used to examine the expression of hundreds or thousands of genes at once, it promises to revolutionize the way scientists examine gene expression. General studies of the expression levels in various kinds of cells represent an important and necessary first step in our understanding and cataloging of the human genome.

As more information accumulates, scientists will be able to use microarrays to ask increasingly complex questions and perform more intricate experiments. With new advances, researchers will be able to infer probable functions of new genes based on similarities in expression patterns with those of known genes. Ultimately, these studies promise to expand the size of existing gene families, reveal new patterns of coordinated gene expression across gene families, and uncover entirely new categories of genes. Furthermore, because the product of any one gene usually interacts with those of many others, our understanding of how these genes coordinate will become clearer through such analyses, and precise knowledge of these inter-relationships will emerge. The use of microarrays may also speed the identification of genes involved in the development of various diseases by enabling scientists to examine a much larger number of genes. This technology will also aid the examination of the integration of gene expression and function at the cellular level, revealing how multiple gene products work together to produce physical and chemical responses to both static and changing cellular needs. The microarray (DNA chip) technology is having a significant impact on genomics study. Drug discovery and toxicological research, will certainly benefit from the use of DNA microarray technology.

3. Principle

The core principle behind microarrays is hybridization between two DNA strands, the property of complementary nucleic acid sequences to specifically pair with each other by forming hydrogen bonds between complementary nucleotide base pairs. A high number of complementary base pairs in a nucleotide sequence means tighter non-covalent bonding between the two strands. After washing off of non-specific bonding sequences, only strongly paired strands will remain hybridized. So fluorescently labeled target sequences that bind to a probe sequence generate a signal that depends on
the strength of the hybridization determined by the number of paired bases, the hybridization conditions (such as temperature), and washing after hybridization. Total strength of the signal, from a spot (feature), depends upon the amount of target sample binding to the probes present on that spot. Microarrays use relative quantization in which the intensity of a feature is compared to the intensity of the same feature under a different condition, and the identity of the feature is known by its position. Many types of array exist and the broadest distinction is whether they are spatially arranged on a surface or on coded beads:

- The traditional solid-phase array is a collection of orderly microscopic "spots", called features, each with a specific probe attached to a solid surface, such as glass, plastics or silicon biochips (commonly known as a genome chip, DNA chip, or gene array). Thousands of them can be placed in known locations on a single DNA microarray.

- The alternative bead array is a collection of microscopic polystyrene beads, each with a specific probe and a ratio of two or more dyes, which do not interfere with the fluorescent dyes used on the target sequence.

4.DEVELOPMENT OF DNA-CHIP

4.1 Combinatorial approach

Multiple sequence alignments reveal conserved and variable sequence regions. This distribution of sequence variability reflects structural and functional constraints. Regions that are tightly constrained are less free to vary than regions which are not as constrained, all else being equal. However, the relationship between functional constraint and variability is not absolute. Some regions can be functionally constrained, but still free to vary for a subset of amino acids that do not jeopardize function. For any given region, there is a finite set of amino acid combinations that can be substituted without changing structure or function. If we could estimate this for all regions of a protein, we could estimate all possible variants viable for a particular protein’s function. This is the principle underlying our gene-specific sequencing chip design. Because ab initio prediction is not yet possible, we use training data sets based on sequences that occur in nature to estimate the sequences that would be viable for a particular protein. A training set comprises a multiple alignment of sequences taken from a diversity of organisms for a given protein-coding gene. The multiple alignment gives an immediate indication of the locations and degree to which sites are free to vary. For example, some sites may be variable but constrained to vary within pyrimidines (C and T), while others may be free to vary across both purine and pyrimidine nucleotides (A, G, C and T). Multiple alignments based on relatively few sequences can provide a surprisingly good indication of the sequence variation that might exist in nature through combinatoric permutation of the observed sequence variation. Consider the two following aligned sequences, variable at positions 1, 4 and 8:

12345678
GAAGCTTA
CAACC

There are eight ways \(2^3\) that the observed differences between the two sequences might be permuted:

12345678
GAAGCTTA
GAAGCTTG
GAACCTTA
GAACCTTG
CAAGCTTA
CAAGCTTG
CAACCTTA
CAACCTTG

The number of combinations implied to be possible expands exponentially as the number of variable sites increases. When this kind of combinatorial expansion is applied to variation that is typical for real data sets, a large number of combinations can result. For example, a pairwise alignment of sequences that vary at 10 sites yields 1024 \((2^{10})\) combinations. A multiple alignment of eight sequences for which 10 sites are variable for all four nucleotides (A, C, G
and T) yields $1048576 (4^{10})$ combinations. A multiple alignment of eight sequences that shows sequence variation restricted to pyrimidine nucleotides at four sites and among all four different nucleotides at six sites would yield $65536 (2^4 \times 4^6)$ combinations. The number of combinatorial variants implied by an alignment of a typical gene 1000 bp in length for which only 10% of sites are variable would be far too large to be represented on a chip. However, if the same multiple alignment is broken down into sections of approximately equal length, say between 15 and 30 nucleotides in length, and the implied variation associated with each particular section is computed separately, the number of variants needed to cover the variation implied by the entire sequence alignment is reduced considerably. By arranging oligonucleotides in a series of columns, each of which corresponds to variation in one section of a gene, we are able to circumvent the sequence reassembly problems that are encountered by many alternative methods. This is the essence of our approach.

5. APPLICATIONS

1. Genomic DNA from an individual is tested for hundreds or thousands of genetic markers in a single hybridization. This will yield a genetic fingerprint, which in turn may be linked to the risk of developing single gene disorders or particular common complex diseases.

2. Sequence variations of specific genes can be screened in a test DNA sample, thereby greatly increasing the scope for precise molecular diagnosis in single gene disorders or complex genetic diseases.

3. DNA microarrays can be used to detect DNA, or detect RNA (most commonly as cDNA after reverse transcription) that may or may not be translated into proteins. The process of measuring gene expression via cDNA is called expression analysis or expression profiling.


5. Identification diseases causing genes using microarray data mining and gene ontology.

6. ADVANTAGES:

1. The DNA chip of Microarray technology shares the problem of standardization with other molecular diagnostic techniques.

2. Accuracy of diagnosis is 85% -90%.

3. Less time required for diagnosis as compared to other molecular diagnostic technique.

4. Identification of genes and hence the associated proteins – that are part of the disease process. Researchers could then use that information to synthesize drugs that interact with these proteins, thus reducing the disease’s effect on the body.

5. Use of this chip may improve treatment results, cut down on side effects and reduce costs potentially spent on a priori useless drugs.

6. An array is a technology that provides massively parallel molecular-genetic information, usually in a visual format.

7. Recent studies demonstrate that chromosomal rearrangements resulting in the loss of genes that play an important role in preventing cells from becoming malignant, a phenomenon known as loss of heterozygosity can be efficiently detected with MicroArrays.

7. Open Access to a View of Systems Biology– All GeneChip microarray content is designed from genomic information in the public domain. The research community has the opportunity to openly collaborate and share data obtained on a single standardized platform. Arrays currently under development show promise of extending these capabilities even further.

8. CONCLUSION

DNA chip technology is the popping up in a varity of application around the world. Each time the technology is applied to biological question, it typically replace much slower, more expensiv method of obtaining answers, or it simply opens up a totally new avenue of injury. DNA microarrays are used for global understanding of abnormal gene expression contributing to malignancy, discovery of new prognostic or predictive indicators and biomarkers of therapeutics response, prediction of drug side effects during preclinical development and toxicology studies, which predicts treatment success or failure. Targeting of therapy may improve treatment results, cut down on side effects and reduce costs potentially spent on a priori useless drugs.

9. REFERENCES:

1. www.gene-chip.com
2. www.cancer.org
3. www.bioinfotech.com
4. www.google.com
5. www.ncbi.com
Vampire Attacks: Draining Life from Wireless Ad-Hoc Sensor Networks


Abstract — Ad-hoc low-power wireless networks are an exciting research direction in sensing and pervasive computing. Prior security work in this area has focused primarily on denial of communication at the routing or medium access control levels. This paper explores resource depletion attacks at the routing protocol layer, which permanently disable networks by quickly draining nodes’ battery power. These “Vampire” attacks are not specific to any specific protocol, but rather rely on the properties of many popular classes of routing protocols. We find that all examined protocols are susceptible to Vampire attacks, which are devastating, difficult to detect, and are easy to carry out using as few as one malicious insider sending only protocol compliant messages. In the worst case, a single Vampire can increase network-wide energy usage by a factor of O(N), where N in the number of network nodes. We discuss methods to mitigate these types of attacks, including a new proof-of-concept protocol that provably bounds the damage caused by Vampires during the packet forwarding phase.

Keywords—Denial of service, security, routing, ad-hoc networks, sensor networks, wireless networks.

I. INTRODUCTION:

Ad-hoc wireless sensor networks (WSNs) promise exciting new applications in the near future, such as ubiquitous-demand computing power, continuous connectivity, and instantly-deployable communication for military and first responders. Such networks already monitor environmental conditions, factory performance, and troop deployment, among a few applications. As WSNs become more and more crucial to the everyday functioning of people and organizations, availability faults become less tolerable — lack of availability can make the difference between business as usual and lost productivity, power outages, environmental disasters, and even lost lives; thus high availability of these networks is a critical property, and should hold even in adverse conditions. Due to their ad-hoc organization, wireless ad-hoc networks are particularly vulnerable to denial of service (DoS) attacks, and a great deal of research has been done to enhance survivability. While these schemes can prevent attacks on the short-term availability of a network, they do not address attacks that affect long-term availability — the most permanent denial of service attack is to entirely deplete nodes’ batteries. This is an instance of a resource depletion attack, with battery power as the resource of interest. In this paper we consider how routing protocols, even those designed to be secure, lack protection from these attacks, which we call Vampire attacks. Ad-hoc wireless sensor networks (WSNs) promise exciting new applications in the near future, such as ubiquitous-demand computing power, continuous connectivity, and instantly-deployable communication for military and first responders. Such networks already monitor environmental conditions, factory performance, and troop deployment, among a few applications. As WSNs become more and more crucial to the everyday functioning of people and organizations, availability faults become less tolerable — lack of availability can make the difference between business as usual and lost productivity, power outages, environmental disasters, and even lost lives; thus high availability of these networks is a critical property, and should hold even in adverse conditions. Due to their ad-hoc organization, wireless ad-hoc networks are particularly vulnerable to denial of service (DoS) attacks, and a great deal of research has been done to enhance survivability. While these schemes can prevent attacks on the short-term availability of a network, they do not address attacks that affect long-term availability — the most permanent denial of service attack is to entirely deplete nodes’ batteries. This is an instance of a resource depletion attack, with battery power as the resource of interest. In this paper we consider how routing protocols, even those designed to be secure, lack protection from these attacks, which we call Vampire attacks, since they drain the life from networks nodes.

Protocols and assumptions: In this paper we consider the effect of Vampire attacks on link-state, distance-vector, source routing, and geographic and...
beacon routing protocols, as well as a logical ID-based sensor network routing protocol proposed by Parno et al. While this is by no means an exhaustive list of routing protocols which are vulnerable to Vampire attacks, we view the covered protocols as an important subset of the routing solution space, and stress that our attacks are likely to apply to other protocols.

All routing protocols employ at least one topology discovery period, since ad-hoc deployment implies no prior position knowledge. Limiting ourselves to immutable but dynamically organized topologies, as in most wireless sensor networks, we further differentiate on-demand routing protocols, where topology discovery is done at transmission time, and static protocols, where topology is discovered during an initial setup phase, with periodic re-discovery to handle rare topology changes. Our adversaries are malicious insiders and have the same resources and level of network access as honest nodes. Furthermore, adversary location within the network is assumed to be fixed and random, as if an adversary corrupts a number of honest nodes before the network was deployed, and cannot control their final positions. Note that this is far from the strongest adversary model; rather this configuration represents the average expected damage from Vampire attacks. Intelligent adversary placement or dynamic node compromise would make attacks far more damaging. While for the rest of this paper we will assume that a node is permanently disabled once its battery power is exhausted, let us briefly consider nodes that recharge their batteries in the field, using either continuous charging or switching between active and recharge cycles. In the continuous charging case, power-draining attacks would be effective only if the adversary is able to consume power at least as fast as nodes can recharge. Assuming that packet processing drains at least as much energy from the victims as from the attacker, a continuously recharging adversary can keep at least one node permanently disabled at the cost of its own functionality. However, recall that sending any packet automatically constitutes amplification, allowing few Vampires to attack many honest nodes. We will show later that a single Vampire may attack every network node simultaneously, meaning that continuous recharging does not help unless Vampires are more resource constrained than honest nodes. Dual-cycle networks (with mandatory sleep and awake periods) are equally vulnerable to Vampires during active duty as long as the Vampire’s cycle switching is insync with other nodes. Vampire attacks may be weakened by using groups of nodes with staggered cycles; only active-duty nodes are vulnerable while the Vampire is active; nodes are safe while the Vampire sleeps. However, this defense is only effective when duty cycle groups outnumber Vampires, since it only takes one Vampire per group to carry out the attack.

Overview:
In the remainder of this paper, we present a series of increasingly damaging Vampire attacks, evaluate the vulnerability of several example protocols, and suggest how to improve resilience. In source routing protocols, we show how a malicious packet source can specify paths through the network which are far longer than optimal, wasting energy at intermediate nodes who forward the packet based on the included source route. Lastly, we show how an adversary can target not only packet forwarding, but also route and topology discovery phases — if discovery messages are flooded, an adversary can, for the cost of a single packet, consume energy at every node in the network. In our first attack, an adversary composes packets with purposely introduced routing loops. We call it the carousel attack, since it sends packets in circles as shown in Figure 1(a). It targets source routing protocols by exploiting the limited verification of message headers at forwarding nodes, allowing a single packet to repeatedly traverse the same set of nodes. Brief mentions of this attack can be found in other literature, but no intuition for defense nor any evaluation is provided. In our second attack, also targeting

(a) An honest route would exit the loop immediately from node E to Sink, but a malicious packet makes its way around the loop twice more before exiting. Fig 1. Malicious route construction attacks on source routing: carousel attack (a)

An example is illustrated in Figure 1(b). Results show that in a randomly-generated topology, a single attacker can use a carousel attack to increase energy consumption by as much as a factor of 4, while stretch attacks increase energy usage by up to an order of magnitude, depending on the position of the malicious node. The impact of these attacks can be further increased by combining them, increasing the number of adversarial nodes in the network, or simply sending more packets. Although in networks that do not employ authentication or only use end-to-end authentication, adversaries are free to replace routes in any overheard packets, we assume that only messages
originated by adversaries may have maliciously-composed routes, damage from Vampire attacks, and find that while the carousel attack is simple to prevent with negligible overhead, the stretch attack is far more challenging. The first protection mechanism we consider is loose source routing, where any forwarding node can reroute the packet if it knows a shorter path to the destination. Unfortunately, this proves to be less efficient than simply keeping global network state at each node, defeating the purpose of source routing. In our second attempt, we modify the protocol to guarantee that a packet makes progress through the network. We call this the no-backtracking property, since it holds if and only if a packet is moving strictly closer to its destination with every hop, and it mitigates all mentioned Vampire attacks with the exception of malicious flooded discovery, which is significantly harder to detect or prevent. We propose a limited topology discovery period (“the night,” since this is when vampires are most dangerous), followed by a long packet forwarding period during which adversarial success is provably bounded. We also sketch how to further modify the protocol to detect Vampires during topology discovery and evict them after the network converges (at “dawn”).

(b) Honest route is dotted while malicious route is dashed. The last link to the sink is shared.

II. RELATED WORK:

We do not imply that power draining itself is novel, but rather that these attacks have not been rigorously defined, evaluated, or mitigated at the routing layer. A very early mention of power exhaustion can be found in [68], as “sleep deprivation torture.” As per the name, the proposed attacks prevent nodes from entering a low-power sleep cycle, and thus deplete their batteries faster. Newer research on “denial of sleep” only considers attacks at the medium access control (MAC) layer. Additional work mentions resource exhaustion at the MAC and transport layers but only offers rate limiting and elimination of insider adversaries as potential solutions. Malicious cycles (routing loops) have been briefly mentioned, but no effective defenses are discussed other than increasing efficiency of the underlying MAC and routing protocols or switching away from source routing. Even in non-power-constrained systems, depletion of resources such as memory, CPU time, and bandwidth may easily cause problems. Less energy to transmit and receive packet. Attackers will produce packets which traverse more hops than necessary, so even if nodes spend the minimum required energy to transmit packets, each packet is still more expensive to transmit in the presence of Vampires. Our work can be thought of as attack-resistant minimal-energy routing, where the adversary’s goal includes decreasing energy savings. Deng et al. discuss path-based DoS attacks and defenses in, including using one-way hash chains to limit the number of packets sent by a given node, limiting the rate at which nodes can transmit packets. Show how protocol-compliant malicious intermediaries using intelligent packet-dropping strategies can significantly degrade performance of TCP streams traversing those nodes. Our adversaries are also protocol-compliant in the sense that they use well-formed routing protocol messages. However, they either produce messages when honest nodes would not, or send packets with protocol headers different from what an honest node would produce in the same situation. Another attack that can be thought of as path-based is the wormhole attack, first introduced in. It allows two non-neighboring malicious nodes with either a physical or virtual private connection to emulate a neighbor relationship, even in secure routing systems. These links are not made visible to other network members, but can be used by the colluding nodes to privately exchange messages. Similar tricks can be played using directional antennas. These attacks deny service.

Fig. 2. Node energy distribution under various attack scenarios. The network is composed of 30 nodes and a single randomly positioned Vampire. Result shown are based on a single packet sent by the attacker.

III. ATTACKS ON STATELESS PROTOCOLS

Here we present simple but previously neglected attacks on source routing protocols, such as DSR [35]. In these systems, the source node specifies the entire route to a destination within the packet header, so intermediaries do not make independent forwarding decisions, relying rather on a route specified by the source. To forward a message,
the intermediate node finds itself in the route (specified in the packet header) and transmits the message to the next hop. The burden is on the source to ensure that the route is valid at the time of sending, and that every node in the route is a physical neighbour of the previous route hop. This approach has the advantage of requiring very little forwarding logic at intermediate nodes, and allows for entire routes to be sender-authenticated using digital signatures, as in Ariadne. We evaluated both the carousel and stretch attacks (Figure 1(a)) in a randomly-generated 30-node topology and asingle randomly-selected malicious DSR agent, using the ns-2 network simulator. In other words, malicious nodes are not driving down the cumulative energy of the network purely by their own use of energy. Nevertheless, malicious node energy consumption data is omitted for clarity. The attacks are carried out by a randomly-selected adversary using the least intelligent attack strategy to obtain average expected damage estimates. More intelligent adversaries using more information about the network would be able to increase the strength of their attack by selecting destinations designed to maximize energy usage. Per-node energy usage under both attacks is shown in Figure 2. Carousel attack. In this attack, an adversary sends a packet with a route composed as a series of loops, such that the same node appears in the route many times. This strategy can be used to increase the route length beyond the number of nodes in the network, only limited by the number of allowed entries in the source route. An example of this type of route is in Figure 1(a). In Figure 3(b), malicious node 0 carries out a carousel attack, sending a single message to node 19 (which does not have to be malicious. On average, a randomly-located carousel attacker in our example topology can increase network energy consumption by a factor of $1.48 \pm 0.99$. The reason for this large standard deviation is that the attack does not always increase energy usage — the length of the adversarial path is multiple of the honest path, which is in turn, affected by the position of the adversary in relation to the destination, so the adversary’s position is important to the success of this attack. Stretch attack. Another attack in the same vein is the stretch attack, where a malicious node constructs artificially long source routes, causing packets to traverse a larger than optimal number of nodes. An honest source would select the route Source → F → E → Sink, affecting four nodes including itself, but the malicious node selects a longer route, affecting all nodes in the network. These routes cause nodes that do not lie along the honest route to consume energy by forwarding packets they would not receive in honest scenarios. An example of this type of route is in Figure 1(b). The outcome becomes clearer when we examine Figure 3(c) and compare to the carousel attack. While the latter uses energy at the nodes who were already in the honest path, the former extends the consumed energy “equivalence lines” to a wider
consumption increase is not as drastic as in (b), the region of increased energy consumption is larger. Overall, energy consumption is greater than in the carousel attack, but spread more evenly over more network nodes.

Fig. 3. Energy map of the network in terms of fraction of energy consumed per node. Black arrows show the packet path through the network. Each dotted line represents an “energy equivalence zone,” similar to an area of equal elevation on a topological chart. Each line is marked with the energy loss by a node as a fraction of total original charge.

IV. ATTACKS ON STATEFUL PROTOCOLS

We now move on to stateful routing protocols, where network nodes are aware of the network topology and its state, and make local forwarding decisions based on that stored state. Two important classes of stateful protocols are link-state and distance-vector. In link-state protocols, such as OLSR, nodes keep a record of the up-or-down state of links in the network, and flood routing updates every time a link goes down or a new link is enabled. Distance-vector protocols like DSDV keep track of the next hop to every destination, indexed by a route cost metric, e.g., the number of hops. In this scheme, only routing updates that change the cost of a given route need to be propagated. Routes in link-state and distance-vector networks are built dynamically from many independent forwarding decisions, so adversaries have limited power to affect packet forwarding, making these protocols immune to carousel and stretch attacks. While this may seem benign in a dense obstacle-free topology, worst-case bounds are no better than in the case of the stretch attack on DSR. For instance, consider the special case of a ring topology: forwarding a packet in the reverse direction causes it to traverse every node in the network (or at least a significant portion of the network), increasing network-wide energy consumption by a factor. It can be performed more than once, depositing the packet at various distant points in the network, at the additional cost to the adversary for each use of the directional antenna. An malicious node has a number of ways to induce a perceived topology change: it may simply falsely claim that a link is down, or claim a new link to a non-existent node. Security measures, such as those proposed by Reffo et al., may suffice to alleviate this particular problem. Therefore, let us assume closed (Sybil-resistant) networks where link states are authenticated, similar to route authentication in Ariadne or path-vector signatures in. Adding more malicious nodes to the mix increases the number of possible router announce/withdrawal pairs. Packet Leashes [30] cannot prevent this attack, with the reasoning being similar to the directional antenna attack — since the originators are themselves malicious, they would forward messages through the wormhole, and return only seemingly valid (and functional) routes in response to discovery.

V. CLEAN-SLATE SENSOR NETWORK ROUTING

In this section we show that a clean-slate secure sensor network routing protocol by Parno, Luk, Gaustad, and Perrig (“PLGP” from here on) [53] can be modified to provably resist Vampire attacks during the packet forwarding phase. The original version of the protocol, although designed for security, is vulnerable to Vampire attacks. PLGP consists of a topology discovery phase, followed by a packet forwarding phase, with the former optionally repeated on a fixed schedule to ensure that topology information stays current. Throughout this process, nodes build a tree of neighbor relationships and group membership that will later be used for addressing and routing. All leaf nodes in the tree are peer nodes in the network, and their virtual addresses correspond to their position in the tree (see Figure 6). All nodes learn each other’s virtual addresses and cryptographic keys. The final address tree is verifiable after network convergence, and all forwarding decisions can be independently verified. Furthermore, assuming each legitimate network node has a unique certificate of membership (assigned before network deployment), nodes who attempt to join multiple groups, produce clones of themselves in multiple locations, or otherwise cheat during discovery can be identified and evicted.

VI. PROVABLE SECURITY AGAINST VAMPIRE ATTACKS

Here we modify the forwarding phase of PLGP to provably avoid the above-mentioned attacks. First we introduce the non-backtracking property, satisfied for a given packet if and only if it consistently makes progress toward its destination in the network address space. More formally: Definition 1. Non-backtracking is satisfied if every packet traverses the same number of hops whether or not an adversary is present in the network. (Maliciously-induced route stretch is bounded to a factor of 1.) This does not imply that every packet in the network must travel the same number of hops regardless of source or destination, but rather that a packet sent to node D by an malicious node at location L will traverse the same number of hops as a packet sent to D by a node at location L that is honest. The only notable exceptions are when adversaries drop or mangle packets en route, but since we are only concerned with packets initiated by adversaries, we can safely ignore this situation: “pre-mangled” packets achieve the same result — they will be dropped by an honest intermediate or destination. No-backtracking implies Vampire resistance. It is not immediately obvious why no-backtracking prevents Vampire attacks in the forwarding phase. Recall the reason for the success of the stretch attack: intermediate nodes in a source route cannot check whether the source-defined route is optimal, or even that it makes progress toward the destination. When nodes make independent routing decisions such as link-state, distance-vector, coordinate-based, or beacon-aided protocols,
packets cannot contain maliciously composed routes. This already means the adversary cannot perform carousels or stretch attacks — no node may unilaterally specify a suboptimal path through the network. However, a sufficiently clever adversary may still influence packet progress. We can prevent this interference by independently checking on packet progress: if nodes keep track of route “cost” or metric and, when forwarding a packet, communicate the local cost to the next hop, that next hop can verify that the remaining route cost is lower than before, and therefore the packets making progress toward its destination. PLGPa differs from other protocols in that packets paths are further bounded by a tree, forwarding packets along the shortest routethrough the tree that is allowed by the physical topology. In other words, packet paths are constrained both by physical neighbour relationships and the routing tree. To preserve no-backtracking, we add a verifiable path history to every PLGP packet, similar to route authentications in Ariadne [20] and path-vector signatures in the resulting protocol, PLGP with attestations (PLGPa) uses this packet history together with PLGP’s tree routing structures so every node can securely verify progress, preventing any significant adversarial influence on the path taken by any packet which traverses at least one honest node. Our adversary is assumed to control $m$ nodes in an $N$-node network (with their corresponding identity certificates and other secret cryptographic material) and has perfect knowledge of the network topology. Finally, the adversary cannot affect connectivity between any two honest nodes.

VII. PERFORMANCE CONSIDERATIONS

PLGP imposes increased setup cost over BVR [21], but compares favorably to in terms of packet forwarding overhead.

While path stretch increases by a factor of 1.5–2, message delivery success without resorting to localized flooding is improved: PLGP never floods, while BVR must flood 5–10% of packets depending on network size and topology [53]. PLGPa also demonstrates more equitable routing load distribution and path diversity than BVR. Since the forwarding phase should last considerably longer than setup, PLGP offers performance comparable to BVR in the average case. PLGPa includes path attestations, increasing the size of every packet, incurring penalties in terms of bandwidth use, and thus radio power. Adding extra packet verification requirements for intermediate nodes also increases processor utilization, requiring time and additional power. Of course, there is nothing to be gained in completely non-adversarial environments, but in the presence of even a small number of malicious nodes, the increased overhead becomes worthwhile when considering the potential damage of Vampire attacks. The bandwidth overhead of our attestation scheme is minimal, as chain signatures are compact (less than 30 bytes). Comparatively, a minimum-size DSR route request packet with no route, payload, or additional options is 12 bytes, whereas 512-byte data packets in our simulations. The additional bandwidth, therefore, is not significant, increasing per-packet transmit power by about 4.8uJ, plus roughly half for additional power required to receive. Energy expenditure for cryptographic operations at intermediate hops is, unfortunately, much greater than transmit or receive overhead, and much more dependent on the specific chipset used to construct the sensor. However, we can make an educated guess about expected performance and power costs. Highly-optimized software-only implementations of AES-128, a common symmetric cryptographic primitive, require about 10 to 15 cycles per byte of data on modern 32-bit x86 processors without AES-specific instruction sets or cryptographic coprocessors. Due to the rapid growth in the mobile space and increased awareness of security requirements, there has been significant recent work in evaluating symmetric and asymmetric cryptographic performance on inexpensive and low-power devices. Their circuitry uses 400 to 800 cycles per round (on 8- and 16-bit architectures, respectively) in the high-current configuration (comparable in terms of clock cycles to AES for RFID [20], but with half to one-tenth the gates and vastly less power), and 1088 cycles when using about 6 times less current. Chain signatures are a somewhat more exotic construction, M and require bilinear maps, potentially requiring even more costly computation than other asymmetric cryptosystems. Assuming a node performs both signature verification as well as a signature append operation, adding attestations to PLGP introduces roughly the same overhead as increasing packet sizes by 90 bytes, taking into account transmit power and cryptographic operations. Without specialized hardware, we estimate cryptographic computation overhead, and thus increased power utilization, of a factor of 2–4 per packet on 32-bit processors, but mostly independent of the route length or the number of nodes in the network: while the hop record and chain signature do grow, their size increase is negligible. In other words, the overhead is constant $O(1)$ for a given network configuration (maximum path length), and cannot be influenced by an adversary. Fortunately, hardware cryptographic accelerators are increasingly common and inexpensive to compensate for increased security demand on low-power devices, which lead to increased computational load and reduced battery life. In total, the overhead on the entire network of PLGP (over PLGP) when using 32-bit processors or dedicated cryptographic accelerator is the energy equivalent of 90 additional bytes per packet, or a factor $O(\lambda x)$, where $\lambda$ is the path length between source and destination and $x$ is 1.2–7.5, depending on average packet size (512 and 12 bytes, respectively).

VIII. SECURING THE DISCOVERY PHASE

Without fully solving the problem of malicious topology discovery, we can still mitigate it by forcing
synchronous discovery and ignoring discovery messages during the intervening periods. This can lead to some nodes being separated from the network for a period of time, and is essentially a form of rate limiting. Although we rejected rate limiting before, its acceptable here since discovery should consume a small fraction of running time compared to packet forwarding. We can enforce rate limits in a number of ways, such as neighbour throttling [35] or one-way hash chains [14]. We can also optimise discovery algorithms [32] to minimize our window of vulnerability. If a network survives the high-risk discovery period, it is unlikely to suffer serious damage from Vampires during normal packet forwarding. While PLGPa is not vulnerable to Vampire attacks during the forwarding phase, we cannot make the same claim about discovery. However, we can give some intuition as to how far we need to modify PLGPa to bound damage from malicious discovery. (The value of that bound in practice remains an open problem.) The major issue is that malicious nodes can use directional antennas to masquerade neighbours to any or all nodes in the network, and therefore look like a group of size one, with which other groups will try to preferentially merge. Messages are composed of the requested group’s ID as well as all the group members’ IDs, and the receiving node will flood this request to other group members. Even assuming groups generate signed tokens that cost no energy to verify, a Vampire would be able to flood its group with every group descriptor it knows, and use its directional antenna to snoop on broadcasts outside their neighbour range, relaying merge requests from entirely honest groups. Since each Vampire will start as a group of one, other groups will issue merge requests, which the Vampire can deny. In PLGPa, denials are only allowed if another merge is in progress, so if we modify the protocol to include the ID of the group with which the merge is in progress (and a signature for non-repudiation), these messages can be kept and replayed at the end of the topology discovery period, detecting and removing nodes who incorrectly deny merge requests. Therefore, Vampires reject legitimate merge requests at their own peril. Any group containing a Vampire can be made to serially join with a “group” composed only of each Vampire in the network (allowing them to advertise themselves as neighbours of each group). Even wholly honest groups can be fooled using directional antennas: Vampires could maintain the illusion that it is a neighbour of a given group. Since join events require multiparty computation and are flooded throughout the group, this makes for a fairly effective attack. PLGPa already provides for the discovery of such subterfuge upon termination of topology discovery: a node who is a member of multiple groups will be detected once those groups join (and all groups are guaranteed to merge by the end of the protocol). Since PLGPa offers the chance to detect active Vampires once the network converges, successive re-discovery periods become safer. This is more than can be said of other protocols, where malicious behaviour during discovery may go undetected, or at least unpunished. However, the bound we can place on malicious discovery damage in PLGPa is still unknown. Moreover, if we can conclude that a single malicious node causes a factor of $k$ energy increase during discovery (and is then expelled), it is not clear how that value scales under collusion among multiple malicious nodes.

IX. CONCLUSION

In this paper we defined Vampire attacks, a new class of resource consumption attacks that use routing protocols to permanently disable ad-hoc wireless sensor networks by depleting nodes’ battery power. These attacks do not depend on particular protocols or implementations, but rather expose vulnerabilities in a number of popular topology classes. We showed a number of proof-of-concept attacks against representative examples of existing routing protocols using a small number of weak adversaries, and measured their attack success on a randomly-generated topology of 30 nodes. Simulation results show that depending on the location of the adversary, network energy expenditure during the forwarding phase increases from between 50 to 1,000 percent. Theoretical worst-case energy usage can increase by as much as a factor of $O(N)$ per adversary per packet, where $N$ is the network size. We proposed defenses against some of the forwarding-phase attacks and described PLGPa, the first sensor network routing protocol that provably bounds damage from Vampire attacks by verifying that packets consistently make progress toward their destinations. We have not offered a fully satisfactorily solution for Vampire attacks during the topology discovery phase, but suggested some intuition about damage limitations possible with further modifications to PLGPa. Derivation of damage bounds and defenses for topology discovery, as well as handling mobile networks, is left for future work.

REFERENCES:
ABSTRACT:
Everybody has the experience of talking aloud in the cell phone in the midst of the disturbance while travelling in trains or buses. There is no need of shouting anymore for this purpose. ‘Silent sound technology’ is the answer for this problem. The Silent sound technology is an amazing solution for those who had lost their voice but wish to speak over phone. It is developed at the Karlsruhe Institute of Technology and you can expect to see it in the near future. When demonstrated, it seems to detect every lip movement and internally converts the electrical pulses. Sounds signals and sends them neglecting all other surrounding noise. It is definitely going to be a good solution for those feeling annoyed when other speak loud over phone. ‘Silent Sound’ technology aims to notice every movement of the lips and transform them into sounds, which could help people who lose voices to speak, and allow people to make silent calls without bothering others.

INTRODUCTION
Silence is the best answer for all the situations …even your mobile understands! The word Cell Phone has become greatest buzz word in Cellular Communication Industry. There are lots and lots of technology that tries to reduce the Noise pollution and make the environment a better place to live in. I will tell about a new technology known as Silent Sound Technology that will put an end to Noise pollution.

SPEECH
It is a technology that helps you to transmit information without using your vocal cords. This technology aims to notice lip movements & transform them into a computer generated sound that can be transmitted over a phone. Hence person on other end of phone receives the information in audio. A gestural form of human communication exists for the deaf in the form of sign language. Speech in some cultures has become the basis of a written language, often one that differs in its vocabulary, syntax and phonetics from its associated spoken one, a situation called diglossia.

Fig.1:-Vocal Tract which converts electric signals into mechanical vibrations, sends sound to the internal ear through the cranial bones. Likewise, a microphone can be used to record spoken sounds via bone conduction. The first description, in 1923, of a bone conduction hearing aid was Hugo Gernsback’s "Osophone", which he later elaborated on with his "Phonosone".
3. SPEECH SYNTHESIS

Synthesized speech can be created by concatenating pieces of recorded speech that are stored in a database. Systems differ in the size of the stored speech units; a system that stores phones or diphones provides the largest output range, but may lack clarity. Speech synthesis is the artificial production of human speech. A computer system used for this purpose is called a speech synthesizer, and can be implemented in software or hardware.

Concatenative synthesis: Concatenative synthesis is based on the concatenation (or stringing together) of segments of recorded speech. Generally, Concatenative synthesis produces the most natural-sounding synthesized speech.

SOUND TECHNOLOGY

Silence is the best answer for all the situation... even your mobile understands!

The word Cell Phone has become greatest buzz word in Cellular Communication industry. There are lots and lots of technology that tries to reduce the Noise pollution and make the environment a better place to live in. I will tell about a new technology known as Silent Sound Technology that will put an end to Noise pollution. You are in a movie theater or noisy restaurant or a bus etc where there is lot of noise around is big issue while talking on a mobile phone. But in the future this problem is eliminated with “silent sounds”, a new technology that transforms lip movements into a computer-generated voice for the listener at the other end of the phone. It is a technology that helps you to transmit information without using your vocal cords. This technology aims to notice lip movements & transform them into a computer generated sound that can be transmitted over a phone. Hence person on other end of phone receives the information in audio.

The Silent sound technology is a perfect solution for those people who have lost their voice but wish to speak on mobile phones. This technology helps to detect every lip movement and converts the electrical pulses into sounds signals and sends those signals avoiding the surrounding noise which may cause disturbance. This is going to be a good solution for those who have lost their intensity to speak. The main aim of ‘Silent Sound’ technology is to notice every movement of the lips and convert them into sound so that the information can be transferred in audio form, which could allow people to make silent calls without bothering about other people. Rather than making any sound, your handset will transfer the movements your mouth makes which will measure the muscle activity with the help of that handset then convert this into audio speech that the person can hear on the other end of the call on phone. So, it reads your lips. This new technology will be very helpful whenever a person loses his. Voice while speaking and it also allows people to make silent calls without disturbing others. This technology can also be used in the military sectors for sharing secret matters with other people. It provides a PIN number which can be given to a trusted person so that the listener can hear a clear voice on the other end.

- Work of Electromyography

It is a technique which monitors tiny muscular movements & pulses generated by it. The
transducer involved converts the pulses into electric signals.

![Image of ultrasound probe and speaker]

**Fig.5:** Process

- **Electrical Characteristics**
  
  The electrical source is the muscle membrane potential of about -90 mV. Measured EMG potentials range between less than 50 μV and up to 20 to 30 mV, depending on the muscle under observation. Typical repetition rate of muscle motor unit firing is about 7–20 Hz, depending on the size of the muscle (eye muscles versus seat (gluteal) muscles), previous axonal damage and other factors. Damage to motor units can be expected at ranges between 450 and 780 mV.

- **Procedure**
  
  There are two kinds of EMG in widespread use: surface EMG and intramuscular (needle and fine-wire) EMG. To perform intramuscular EMG, a needle electrode or a needle containing two fine-wire electrodes is inserted through the skin into the muscle tissue. A trained professional (such as a neurologist, physiatrist, or physical therapist) observes the electrical activity while inserting the electrode. The insertion activity provides valuable information about the state of the muscle and its innervating nerve. Normal muscles at rest make certain, normal electrical signals when the needle is inserted into them. Then the electrical activity when the muscle is at rest is studied. Abnormal spontaneous activity might indicate some nerve and/or muscle damage. Then the patient is asked to contract the muscle smoothly. The shape, size, and frequency of the resulting motor unit potentials are judged. Then the electrode is retracted a few millimeters, and again the activity is analyzed until at least 10–20 units have been collected.

  Each electrode track gives only a very local picture of the activity of the whole muscle. Because skeletal muscles differ in the inner structure, the electrode has to be placed at various locations to obtain an accurate study.

  Intramuscular EMG may be considered too invasive or unnecessary in some cases. Instead, a surface electrode may be used to monitor the general picture of muscle activation, as opposed to the activity of only a few fibers as observed using an intramuscular EMG. This technique is used in a number of settings; for example, in the physiotherapy clinic, muscle activation is monitored using surface EMG and patients have an auditory or visual stimulus to help them know when they are activating the muscle (biofeedback).

  Nerve conduction testing is also often done at the same time as an EMG to diagnose neurological diseases.

  Some patients can find the procedure somewhat painful, whereas others experience only a small amount of discomfort when the needle is inserted. The muscle or muscles being tested may be slightly sore for a day or two after the procedure.

**10. RESULTS**

a) **Normal results:**

  Muscle tissue at rest is normally electrically inactive. After the electrical activity caused by the irritation of needle
insertion subsides, the electromyography should detect no abnormal spontaneous activity (i.e., a muscle at rest should be electrically silent, with the exception of the area of the neuromuscular junction, which is, under normal circumstances, very spontaneously active). When the muscle is voluntarily contracted, action potentials begin to appear. As the strength of the muscle contraction is increased, more and more muscle fibers produce action potentials. When the muscle is fully contracted, there should appear a disorderly group of action potentials of varying rates and amplitudes (a complete recruitment and interference pattern).

b) Abnormal results:
EMG is used to diagnose diseases that generally may be classified into one of the following categories: neuropathies, neuromuscular junction diseases and sympathies. Neuropathic disease has the following defining EMG characteristics:
- An action potential amplitude that is twice normal due to the increased number of fibers per motor unit because of reinnervation of denervated fibers
- An increase in duration of the action potential
- A decrease in the number of motor units in the muscle (as found using motor unit number estimation techniques)

Myopathic disease has these defining EMG characteristics:
- A decrease in duration of the action potential.
- A reduction in the area to amplitude ratio of the action potential.
- A decrease in the number of motor units in the muscle (in extremely severe cases only).

Because of the individuality of each patient and disease, some of these characteristics may not appear in every case.

11. EMG SIGNAL DECOMPOSITION
EMG signals are essentially made up of superimposed motor unit action potentials (MUAPs) from several motor units. For a thorough analysis, the measured EMG signals can be decomposed into their constituent MUAPs. MUAPs from different motor units tend to have different characteristic shapes, while MUAPs recorded by the same electrode from the same motor unit are typically similar. Notably MUAP size and shape depend on where the electrode is located with respect to the fibers and so can appear to be different if the electrode moves position. EMG decomposition is non-trivial, although many methods have been proposed.

12. ADVANTAGES
- Best suits for astronauts, as there was no medium for communication of sound.
- It is capable of translating into many languages as the electrical signals are universal.
- Silent sound technology has lifted the speech recognition technology to the superior level.
- Reduce noise.

13. DISADVANTAGES
- Though silent sound technology has developed with 99% of accuracy, still it is prone to some errors.
- Regarding the security, it is quite complicated to accommodate.
- Differentiating people across the world and emotions are not recognized.
- The user has a feel that talking to a robot when they receive final output.
- It performs the function of translation but for languages like Chinese has different tone in critical situations it fails to translate with accuracy (ch).
16. FUTURE WORK
The technology can also turn you into an instant polyglot. Because the electrical pulses are universal, they can be immediately transformed into the language of the user’s choice.

“Native speakers can silently utter a sentence in their language, and the receivers hear the translated sentence in their language. It appears as if the native speaker produced speech in a foreign language,” said Wand.

The translation technology works for languages like English, French and German, but for languages like Chinese, where different tones can hold many different meanings, poses a problem, he added. “We are also working on technology to be used in an office environment,” the KIT scientist told AFP.

“But we’re working to overcome the remaining technical difficulties. In five, maybe ten years, this will be useable, everyday technology.

17. CONCLUSION
Thus Silent Sound Technology, one of the recent trends in the field of information Technology implements “Talking without Talking”.

It will be one of the innovation and useful technology and in mere future this technology will be use in our day to day life.

‘Silent Sound’ technology aims to notice every movement of the lips and transform them into sounds, which could help people who lose voices to speak, and allow people to make silent calls without bothering others. Rather than making any sounds, your handset would decipher the movements your mouth makes by measuring muscle activity, then convert this into speech that the person on the other end of the call can hear. So, basically, it reads your lips.

Engineers claim that the device is working with 99 percent efficiency.

It is difficult to compare SSI technologies directly in a meaningful way. Since many of the stems are still preliminary, it would not make sense, for example, to compare speech recognition scores or synthesis quality at this stage.

With a few abstractions, however, it is possible to shed light on the range of applicability and the potential for future commercialization of the different methods.

18. REFERENCES


ABSTRACT
Transcatheter Aortic Valve Implantation (TAVI) is a new therapy which may be used as an alternative to standard surgical aortic valve replacement. The procedure is performed on the beating heart without the need for a sternotomy or cardiopulmonary bypass and involves replacing a diseased aortic valve, usually one with narrowing or stenosis, with a new valve. Transcatheter aortic valve implantation (TAVI) is becoming a reality in the management of patients with severe aortic stenosis and high or prohibitive risk for standard surgical management. Risks associated with TAVI differ from those related to surgical valve replacement and include vascular injury; stroke; cardiac injury such as heart block, coronary obstruction, and cardiac perforation; paravalvular leak; and valve misplacement. The clinical experience of multiple centers experience with different valve implantation systems and techniques was reviewed. Ultimately, improved understanding of the potential complications associated with TAVI might help improve outcomes and allow wider application of this therapy. Current understanding of the potential adverse events associated with this procedure is limited.

Keywords: Heart valve, Aortic stenosis, Edwards Sapien Valve, Medtronic Core Valve.

1. INTRODUCTION
The aortic valve is one of four valves in the human heart (Figure 1.1). It is located between the left ventricle and aorta and in 99 percent of individuals is tri-leaflet in structure (in 1 percent of cases it can be bicuspid or unicusp ). During left ventricular systole (contraction), the pressure in the left ventricle increases until it rises just above the systolic pressure in the aorta. At this point in systole, the aortic valve opens and blood exits the left ventricle into the systemic circulation via the aorta. Thereafter, during left ventricular diastole (relaxation), the pressure in the left ventricle drops, and the pressure in the aorta forces the aortic valve back into its closed position. There are two primary disease processes that can affect the aortic valve: aortic insufficiency and aortic stenosis. In aortic insufficiency, also referred to as aortic regurgitation, the aortic valve is in competent or leaky, and blood flows back into the left ventricle from the aorta during diastole. In aortic stenosis, the valve fails to open fully, thereby creating a systolic pressure gradient between the left ventricle and aorta. Demonstrating the four chambers (left and right atria, left and right ventricles) the four main valves (mitral, tricuspid, aortic, and pulmonary).

For many of these older patients, surgical aortic valve replacement (SAVR) is a high risk procedure with significant morbidity and mortality. To meet the medical needs of this population, a new technology has emerged over the past decade and is now being put into clinical practice: transcatheter aortic valve replacement.

This technology has had significant impacts throughout the health care field with the creation of a new biomedical industry around transcatheter valves, the creation of multidisciplinary “heart teams” within clinical practice, and the construction of hybrid procedure rooms with both cathlab and operating room capabilities.
2. DESIGN OF TRANSCATHETER AORTIC VALVES:

Both the Edwards-Sapien Valve and the Medtronic CoreValve are designed to function through a mechanism similar to a normally functioning human tricuspid aortic valve. However, while both are tri-leaflet in design with a metallic framework for support, their construction as well as preparation and delivery have significant differences. The integrated Edwards-Sapien transcatheter heart valve system is comprised of bovine pericardial tissue made from three identical sections of bovine pericardium that have been preserved in buffered glutaraldehyde to enable crosslinking of the tissue while preserving flexibility and strength. The valve tissue is affixed to a radiopaque stainless steel stent frame within a fabric cuff at its inflow aspect and to attachment bars on the commissural posts at its outflow aspect using polytetrafluoroethylene (PTFE) sutures to create a unidirectional trileaflet tissue valve.

A. Fully expanded Edwards Sapien Valve with its stainless steel frame and trileaflet construction made from bovine pericardial tissue.

B. Fully expanded Medtronic CoreValve with its nitinol frame and trileaflet construction made from porcine pericardial tissue

3. METHODOLOGY

3.1 Working:

The Edwards SAPIEN heart-valve system (Edwards Life sciences) consists of a trileaflet bovine pericardial valve and a balloon-expandable, stainless steel support frame. The heart valve is shown in fig. The TAVI procedure was performed in a sterile environment (catheterization laboratory or operating room), with the patient under general anesthesia; the procedure was performed with the use of transesophageal echocardiography. A standard balloon aortic valvuloplasty was performed, followed by transfemoral insertion of either a 22- or 24-French sheath, depending on the selected size of the valve (23 mm or 26 mm). The bioprosthetic heart valve, crimped onto a balloon catheter, was advanced across the native aortic valve. During rapid right ventricular pacing, balloon inflation of the crimped heart valve and support frame simultaneously deployed the bioprosthetic valve and expanded the frame, which was secured to the underlying aortic-valve annulus and leaflets.

3.2 There are two common routes of new valve insertion:

1. Transfemoral - through the femoral artery, the main artery in your groin which leads back to the heart.

2. Transapical - through a small cut on the left side of your chest to get to the apex (tip) of your heart.

4. PROCEDURE

A standard balloon aortic valvuloplasty was performed, followed by transfemoral insertion of either a 22- or 24-French sheath, depending on the selected size of the valve (23 mm or 26 mm).

The bioprosthetic heart valve, crimped onto a balloon catheter, was advanced across the native aortic valve. During rapid right ventricular pacing, balloon inflation of the crimped heart valve and support frame simultaneously deployed the bioprosthetic valve and expanded the frame, which was secured to the underlying aortic-valve annulus and leaflets. Adjunctive pharmacologic therapy included heparin during the procedure and dual antiplatelet therapy (aspirin and clopidogrel) for 6 months after the procedure.
4.1 What the procedure involves?

TAVI aims to provide a less invasive alternative to open cardiac surgery for the treatment of aortic stenosis, avoiding the need for cardiopulmonary bypass. The procedure is carried out under general anaesthesia or using local anaesthesia with sedation. Imaging guidance, including fluoroscopy, angiography and transoesophageal echocardiography is required. Prophylactic antibiotics and anticoagulation medication are administered before and during the procedure. Temporary peripheral extracorporeal circulatory support (usually via the femoral vessels) is sometimes used. The procedure implants a bioprosthetic aortic valve at the site of the native aortic valve. Access to the aortic valve can be achieved transluminally, with entry to the circulation usually achieved via the femoral or other large artery or vein (sometimes known as a percutaneous, or endovascular approach); or surgically, with access to the aortic valve via apical puncture of the left ventricle using a minithoractomy approach (transapical, or transventricular approach). In the transluminal approach, when the femoral vein is used, the interatrial septum is punctured in order to gain access to the left ventricle via the left atrium and mitral valve; when the femoral or other large artery is used, surgical exposure and closure may be required. The choice of how catheter access to the aortic valve is achieved may depend on the existence of factors that make passage through the circulation difficult such as peripheral vascular disease.

Manipulated into position and placed over the existing aortic valve. In order to provide a stable platform for aortic valve implantation, rapid right ventricular pacing is used to temporarily interrupt blood flow through the native aortic valve.

The new valve is mounted on a metal stent which is either self-expanding or expanded using inflation of a large balloon on which the stented valve has been crimped. Positioning the new valve leads to obliteration of the native aortic valve. The delivery catheter is removed after successful valve placement.

Fig.4.1 Insertion of Catheter

Fig.4.2: Angiography

Antibiotics and anticoagulation medication are administered before and during the procedure. Temporary peripheral extracorporeal circulatory support (usually via the femoral vessels) is sometimes used. The procedure

Fig.4.2(a): The delivery system carrying the valve guided by a type of X-ray.

Fig.4.2(b): The balloon of the delivery system carrying

Fig.4.2(c): Your doctor will make sure that your new valve is working proper.
5. Conclusions:

- New recommendations and expert consensus documents are being developed to help guide the use of this new technology.
- We are no longer in an era in which the old boundaries between cardiac specialties apply.
- To obtain the best outcome for the patient, there must be a true collaboration between the cardiothoracic surgeon and interventional cardiologist, not only in evaluation of the patient but in the procedure itself.
- Just as with the development of the valve design, the development of the “heart valve team” will take time but is just as critical for the success of this technology.

REFERENCES:

PSO Based Adaptive Filter for Unknown System Identification


Department of Electronics & Telecommunication Engineering
DBNCOET Yavatmal, Maharashtra India

Abstract— Particle swarm optimization is a heuristic global optimization method and also an optimization algorithm, which is based on swarm intelligence. It comes from the research on the bird and fish flock movement behaviour. The algorithm is widely used and rapidly developed for its easy implementation and few particles required to be tuned. This paper focuses on the implementation of Adaptive Infinite Impulse response (IIR) filter using Modified Particle Swarm Optimization (PSO) Algorithm.

The application of particle swarm optimization (PSO) technique to find the optimal location of flexible AC transmission system (FACTS) devices with minimum cost of installation of FACTS devices and to improve system load ability (SL). While finding, size, optimal location, thermal limit for the lines and voltage limit for the buses are taken as FACTS devices used as thyristor controlled series compensator (TCSC). Structured stochastic optimization algorithms that are effective on multimodal error surfaces are then introduced, with particular attention to the Particle Swarm Optimization (PSO) technique.

Keywords: Adaptive IIR Filter, System Identification, Particle Swarm Optimization.

I. Introduction

Particle swarm optimization is a heuristic global optimization method put forward originally by Doctor Kennedy and Eberhart in 1995(Kennedy J, Eberhart R, 1995; Eberhart Kennedy J, 1995) It is developed from swarm intelligence and is based on the research of bird and fish flock movement behaviour. While searching for food, the birds are either scattered or go together before they locate the place where they can find the food. While the birds are searching for food from one place to another, there is always a bird that can smell the food very well, that is, the bird is perceptible of the place where the food can be found, having the better food resource information. Because they are transmitting the information, especially the good information at any time while searching the food from one place to another, conducted by the good information, the birds will eventually flock to the place where food can be found. As far as particle swarm optimization algorithm is concerned, solution swarm is compared to the bird swarm, the birds’ moving from one place to another is equal to the development of the solution swarm, good information is equal to the most optimist solution, and the food resource is equal to the most optimist solution during the whole course.

Adaptive filters have become vastly popular in the area of digital signal processing. Adaptive direct modelling or system identification and adaptive inverse modelling or channel equalization find extensive applications in telecommunication, control system, instrumentation. Adaptive Infinite Impulse Response (IIR) systems are used in modelling real world systems because of their reduced number of coefficients and better performance over the Finite Impulse Response (FIR) filter.

II. LITERATURE REVIEW

PSO is one of the optimization techniques and belongs to evolutionary computation techniques. PSO is basically developed through simulation of bird flocking in two-dimension space. The position of each individual (agent) agent position is realized by the position and velocity information. An optimization technique based on the above concept can be described as follows: namely, bird flocking optimizes a certain objective function. Each agent knows its best value so far (pbest) and its XY position. Moreover, each agent knows the best value so far in the group (gbest) among pbests agent tries to modify its position. Cognition and social Current position. Based on Darwinian evolutionary theory Population-based Members of the population are called individuals Determine the fitness (quality) of each individual using an objective function Rank population according to fitness Produce new population using best individuals Based on the social interaction exhibited by animals Particle refers to a potential solution Collection of particles is called swarm Population-based Update particles using a cognitive and social model Improve performance using different particle.
III. PARTICLE SWARM OPTIMIZATION

PSO is a robust stochastic optimization technique based on the movement and intelligence of swarms. PSO applies the concept of social interaction to problem solving. It was developed in 1995 by James Kennedy (social-psychologist) and Russell Eberhart (electrical engineer). It uses a number of agents (particles) that constitute a swarm moving around in the search space looking for the best solution.

Each particle keeps track of its coordinates in the solution space which are associated with the best solution (fitness) that has achieved so far by that particle. This value is called personal best, pbest. Another best value that is tracked by the PSO is the best value obtained so far by any particle in the neighbourhood of that particle. This value is called gbest. PSO takes its inspiration from the behaviour of birds, insects and their communities, and how they manage as a group, rather than as individuals, recreating themselves and adapting in accordance with the changes in the surrounding environment, in order to search for food or to migrate. In other words, PSO is mainly inspired by social behaviour patterns of organisms that live and interact within large groups.

Uses of Optimization
In the field of machine learning, optimization techniques can be used to find the parameters for classification algorithms such as:

- Artificial Neural Networks
- Support Vector Machines

These classifications of optimization method & algorithms often require the user to supply certain coefficients, which often have to be found by trial and error or exhaustive search.

The definition of a FACTS device given by the IEEE is "a power electronic base system and other static equipment that provide control of one or more AC transmission and distribution system parameters to enhance controllability and increase power transfer capability". FACTS devices include Thyristor controlled series compensator (TCSC).

- In PSO, there have been two basic topologies used in the literature
  - Ring Topology (neighbourhood of 3)
  - Star Topology (global neighbourhood)

In the basic particle swarm optimization algorithm, particle swarm consists of “n” particles, and the position of each particle stands for the potential solution in D-dimensional space. The particles change its condition according to the following three principles: to keep its inertia to change the condition according to its most optimist position to change the condition according to the swarm’s most optimist position. The position of each particle in the swarm is affected both by the most optimist position during its movement (individual experience) and the position of the most optimist particle in its surrounding (near experience). If the narrow surrounding is used in the algorithm, this algorithm is called the partial PSO. The PSO method is based on swarm intelligence. The mathematic foundation includes the mechanical principle of PSO itself, the prove of its convergence and Robustness and etc. The study on PSO should be concentrated on the following: some modern technologies should be applied to PSO to design the improved PSO; PSO can be combined with the other intelligent optimization methods to design several compound optimization methods; PSO can be also led into scattering system, compound optimist system, non-coordinate system to develop PSO’s application ranges.
IV. ADAPTIVE FILTER

An adaptive filter is a computational device that attempts to model the relationship between two signals in real time in an iterative manner. Adaptive filters are often realized either as a set of program instructions running on an arithmetical processing device such as a microprocessor or DSP chip, we shall focus on the mathematical forms of adaptive filters as opposed to their specific realizations in software or hardware. Descriptions of adaptive filters as implemented on DSP chips and on a dedicated integrated circuit can be found.

An adaptive filter is defined by four aspects:
1. The signal being processed by the filter
2. The structure that defines how the output signal of the filter is computed from its input Signal.
3. The adaptive algorithm that describes how the parameters are adjusted from one time instant to the next.

In this section, we present the general adaptive filtering problem and introduce the mathematical notation for representing the form and operation of the adaptive filter. Selecting the correct order and estimating the parameters of the adaptive filter is a fundamental issue in system identification. In the last decades, substantial effort has been done to use IIR adaptive filtering techniques as an alternative to adaptive FIR filters.

V. PSO based Adaptive IIR Filter

In System identification, it is necessary to filter one signal y (n) in order to match another signal d (n) as closely as possible. Most nonlinear systems are also recursive in nature. Hence, models for real world systems are better represented as IIR systems. By doing so, the problem of system identification now becomes the problem of adaptive IIR filtering, for which different adaptive algorithms can be applied for adjusting the feed forward and feedback parameters of the.

$$y(n) = \sum a_i(n)u(n-i) + \sum b_i(n)y(n-i)$$

Forward filter  backward filter

To build the adaptive process around a linear IIR filter with fewer number of adjustable coefficients than an FIR filter to achieve a desired response two approaches:
1) Output error method
2) Equation error method

VI. CONCLUSION

Particle swarm optimization is seems to be effective for optimizing wide range of functions. We view it as a mid-level form of A-life or biologically derived algorithm, occupying the space in nature between evolutionary searches, which requires neural processing, which occurs on the order of milliseconds. The adjustment toward pbest and gbest by the particle swarm optimizer algorithms. Resource selection based on PSO is used to generate an optimal schedule so as to complete the tasks in a minimum time than PSO as well as utilizing the resources in an efficient way.

REFERENCE

[1] The particle swarm optimization algorithm: convergence analysis and parameter selection Iona Christian Tralee INA P-G, UMR Génie et Microbiologie des Procédés Alimentaires, BP 01, 78850 Thiverval-Grignon, France Received 10 July 2002; received in revised form 12 September 2002
Abstract— Three dimensional network on chip (3D NoC ) is that the most thriving on chip connection design. That is network on chip mesh design has been instructed as answer to handle international communication trials in system on chip (SoC) design. In this literature survey, the performance improvement originating from the architectural advantages of 3D network on chip are considerably increased. By emerging 3D IC accomplish larger device integration and enhance the system presentation at smaller cost and reduces communication distance in 3D NoC. Additionally to its has numerous advantages in terms of power utilization and system performance has chance to implement an efficient design. During this survey, effective 3D network on chip design is usually recommended with the mechanism of congestion aware rule that optimizes the facility utilization, system performance and minimize latency with the help of routing algorithm and topology. In this survey basically we focus on mesh and torus network. In supplement we have integrated the reduced cost platform of 3D NoC design which might be effectively utilized for fault tolerant and decreased traffic. Supported routing scheme for 3D NoC will facilitate to realize significant power utilization and decreased the latency.

Keywords— 3D Integration, 3D Network on Chip, Routing Algorithm, Selection Function, Direct Network Topology.

Introduction
Global interconnects have become the principal performance bottleneck high performance Systems on Chip (SoCs)[1]. A typical bus based systems are not have any longer reliable design for SoC owing to an absence of scalability and correspondence integration, high latency and power dissipation, and low throughput. Network on Chip [1][9][18] is presented as a revolutionary technique that may overcome these issues by presenting an easy and scalable design platform, impressed by the internet. which NoC to produce a better bandwidth and better performance.

In the past few years, three dimensional integrated circuits (3D-ICs) [4] have attracted a lot of attention. Results obtained as far show that 3D-ICs can do higher performances, a lot of flexibility, and better throughput compared to traditional ICs. This could provide a chance to continue performance enhancements using CMOS technology with smaller form factors, higher integration densities, and supporting the conclusion of mixed technology chips [2][3]. As Topol et al in [4] introduced, 3D-IC will improve the performance even in absence of scalability. Apart from this clear benefit of package density is enhanced considerably, power is reduced from use of shorter wires, and circuitry is a lot of immune to noise.

Combining the NoC structure with the advantages of the 3D integration lead us to conclude 3D NoC as a replacement design [4]. This structure responds to the scaling demands for future SoC [1] exploiting the short vertical links between the adjacent layers which will clearly enhance the system performance. As per Feero et al [7], 3D NoC has the flexibility to decrease the quantity of hops and necessary issue to measure the system performance. 3D mesh NoC has close to concerning absolutely less hops than the 2D mesh NoC. This reduction will effectively increase throughput and consequently latency decreases as a result of flits traverse less hops whereas traveling from source to destination. Another crucial issue have to measure the performance of a system is power dissipation. Power dissipation becomes more and more necessary in designing efficient 3D NOC. In [6], results show that 3D NoC uses less energy per packet than 2D NoC implementations. The analysis results concerning 2D and 3D NoC performance are explained in detail in [7][6] and [8]. In term of network topology D. Fick presented the implementations of the algorithm for 2D mesh and 2D torus networks, it presents the solution routes around network failures by leveraging redundancy inherent in on-Chip topologies and also results torus topology for 3D NoC gives better performance than mesh.[10][11].

Then again this survey shows, routing throughout the network, which are used to verify the path of a packet from the source to the destination. These algorithms are classified as deterministic and adaptive. The implementations of deterministic routing algorithm are easy however they are unable to balance the load across the links in non-uniform or bursty traffic [14][5]. Reconciling routing algorithms are design to handle these limitations. By better distributing load across links, adaptive algorithms improve network performance and also provide tolerance if link or router failure happens. In adaptive routing algorithms [9][11], the path of a packet from the source to the destination is decided by the network condition. An adaptive routing algorithmic rule decreases the probability of passing a packet from a congested or mal-function link. Despite its implementation quality, the routing is attractive for large NoCs especially when these NoCs facing with non-uniform or bursty traffic[17][14].
II. THE BASICS OF NOC ARCHITECTURE

A typical NOC consists of computational processing element (PEs), network interfaces (NIs), and routers. The latter two comprise the communication architecture [27]. An NI is employed to packetize information before using the router backbone to traverse the NoC. Each PE is connected an NI and that connects the PE to a local router. When a packet was sent from a source to a destination as shown in Fig. 2.A, the packet is forwarded hop by hop on the network via the choice made by each router. In some NOC architectures that are equipped with error control mechanisms, NIs are also accustomed encode and decode the information by the error control code applied.

![Fig 2 A. Network on chip architecture](image1)

Fig 2 A. Network on chip architecture

An NOC router consists of switches, registers, and control logic that collectively perform routing and channel arbitration to guide the flow of packets in the network as illustrated in Fig. 2.B for every router, the packet is first received and stored in an input buffer. Then the control logic within the router is responsible to make routing decision and channel arbitration. Finally, the granted packet can traverse through a crossbar to destination router, and therefore the method repeats till the packet arrives at its destination.

Packets delivered by routers are partitioned in a flit-by-flit basis. Each flit of a packet arrives at a router and store in a memory buffer until it can traverse to the next hop of the route. The first flit in the buffer memory will be processed by the control logic to determine whether it is allowed to be forwarded and which output direction it should proceed to. The decision made by the control unit is based on the computation result of routing, arbitration, and the downstream buffer space. After the control setup is done, the flit passes through the crossbar switch to its desired output direction.

![Fig 2 B. Typical NoC Router architecture](image2)

Fig 2 B. Typical NoC Router architecture

III. RELATED WORK:

The literature related to this paper is summarized in four sub-sections: 3D Integration technology, 3D Network on Chip Architectures, Topology and Routing Algorithm. These sections offer a short survey on the exploration of attainable architectural designs for three-dimensional NoC architectures and discuss the tradeoffs among varied design choices.

A. 3D Integration Technology

Three-dimensional integration technology [2] is an attractive option for overcoming the barriers in interconnect scaling, offering an opportunity to continue the CMOS performance trend. In a three-dimensional chip, multiple device layers are stacked together. Various 3D integration vertical interconnect technologies have been explored, including wire bonded, micro bump, contactless (capacitive or inductive), and through silicon-via vertical interconnect [2][3]. Through silicon-via interconnection has the potential to offer the greatest vertical interconnect density and therefore is the most promising one among these vertical interconnect technologies. There are two different approaches to implementing through-silicon-via 3D integration: the first one involves sequential device process, in which the front-end processing (to build the device layer) is repeated on a single wafer to build multiple active device layers, before the interconnects among devices are built. The second approach processes each active device layer separately, using conventional fabrication techniques, and then stacking these multiple device layers together using wafer-bonding technology [2].
Thermal considerations have been a significant concern for 3D integration [4]. However, various techniques have been developed to address thermal issues in 3D architectures such as physical design optimization through intelligent placement, increasing thermal conductivity of the stack through insertion of thermal-vias, and use of novel cooling structures [6]. Further, a recent work demonstrated that the real power density is the more important design constraint in placement of the processing cores in a 3D chip, as compared to their location in the 3D stack and presented the several electrical benefits of 3D integration [6]. Consequently, thermal concern can be managed as long as components with high power density are not stacked on top of each other. Architectures that stack memory on top of processor cores, or those that rely on low-power processor cores have been demonstrated to not pose severe thermal problems [4][6]. In spite of all these advances, one can anticipate some increase in temperature as compared to a 2D design and also a temperature gradient across layers. Increased temperatures increase wire resistances, and consequently the interconnect delays. To capture this effect, we study the impact of temperature variations on the 3D interconnect delay to assess the effect on performance.

As per [3][4][6], Three-dimensional integrated circuits (3D ICs) provide an attractive resolution for overcoming the barriers to interconnect scaling, thereby giving a chance to continue performance enhancement using CMOS technology, with smaller type issue, higher integration density, and also the support for the conclusion of mixed-technology chips. Among many 3D integration technologies [2], as shown in fig.2 TSV (Through-Silicon-Via) approach is the most promising one and so is the focus of the majority of 3D integration of several activities, even though each 3D integrated circuits and 3D NoCs [4] are planned as alternatives for the interconnect scaling demands, the challenges of combining both approaches to design three-dimensional NOCs recently.

**B. NoC Topology**

**Title** The topology of a NOC [3] specifies the physical organization of the interconnection network. It defines the interconnection between nodes, switches, and links [2]. One in all the kind of topology configuration in direct network topologies [14], every node has direct point-to-point links to a subset of alternative nodes within the system known as neighboring nodes. The node carries with it computational blocks and/or memories, as well as a network interface (NI) block that acts as a router. This router is connected to the router of the neighboring nodes through links. Most direct network topologies have associated orthogonal NOC topology. Most direct network topologies have associate orthogonal implementation[14], whenever the nodes are organized in associate n dimensional orthogonal area, in such the way that each link produces a displacement in a very single direction. Routing for such networks is fairly easy and may be implemented expeditiously in hardware. Samples of well-liked orthogonal direct network include the n-dimensional mesh, torus, folded torus, hypercube and polygon topologies. Because of simple implementation mostly the mesh and torus topology was used for 3DNoC as shown in fig 4.

**Fig 3. 3D mesh and Torus topology**

The topology of 3D NoC concerns the placement and interconnection of NoC nodes. Protocols specify how these nodes and links work [2]. Several common topology of NoC are introduced in[15]. A form of topologies mainly includes regular and irregular forms[16]. Mesh, Torus, Tree are regular forms of topologies. Irregular forms of topologies are derived by mixing different forms, and a hierarchical, hybrid or asymmetric fashion. Irregular forms of topologies scale non-linearly with regards to area and power, and are complicated in wiring [7]. Of these NoC topologies Mesh and Torus with regularity are mainly used in NoC design and prone to VLSI implementation [8] [15] [17].

Pavlidis et al introduced the many fascinating topologies emerge by incorporating the third dimension in networks-on-chip[8]. The speed and power consumption of 3D NOC are compared to that of 2D NOC. Physical constraints, like the maximum number of planes that may be vertically stacked and also the imbalance between the horizontal and vertical communication channels of the network, are included in speed and power consumption models. Also perform the work on zero load latency and power consumption.

D. Fick [11] presented the comparison result of mesh and torus topology for 2D network on chip. Because of the simple physical implementation of mesh topology mostly used and
also perform the work on torus topology for fault tolerant of 2D NoC [17]. Also Yang Xiaoqiang [17] presented the work on torus topology for the improvement of x-y routing algorithm along the 2D NoC architecture.

The research on torus topology for 3D network on chip proposed in [10]. In this case the performance of mesh topology network degrades the phenomenal increased in diameter hence as a solution, the torus network used for this NoC architecture. A torus topology is same as mesh topology but the difference is that the torus having an extra node i.e. called as wrap around edge which connect the boundary node of mesh network. Hence with wrap around edge which help reduced the diameter of the network [10].

C. 3D Network on Chip architecture

Routing schemes have been classified in several ways in the literature i.e. by using selection function, deterministic or adaptive routing. As per W. feng et al. [24] analyze in detail many selection functions so as to evaluate their influence on network performance. Their simulation results show that network throughput could also be increased. A number of selection functions are projected for adaptive routing algorithms; the best one could also be the random selection function that randomly chooses one output channel from all permissible channels provided by the routing function. Obviously, this selection doesn’t create use of any network data; therefore it doesn’t usually deliver satisfying performance.

Yanbin Lin proposed selection function is predicated on the idea of dynamic-bandwidth-estimation (DBE). With the DBE selection function, every router grasps congestion data locally, estimate the actual bandwidth of every output channel based on the grasped information, and try to forward the packet through the channel with the best estimated bandwidth[18]. It bypassed the congested area and link however needs to improve throughput and latency of system.

G.Ascia have presented a selection strategy that introduces the idea of NoP (neighbors-on-path) to boost the performance of an NoC[26]. The aim of our algorithmic rule is to take advantage of situation of incidence which will occur in adaptive wormhoerouting. The approach, that is general in nature, has been applied to the OE and compared to each deterministic and adaptive routing algorithm. The simulations performed showed, in most of the cases, an improvement in average delay and energy consumption, particularly under serious traffic workloads. The comparison of routing and selection ways strictly depends on the traffic scenario that populates the NOC. But, developments include totally different interesting issues, such as the analysis of various topologies and traffic patterns, an additional and essential step are going to be the mapping of real applications on NoCs, and also to improve choice of routing strategy for that higher packet injection rates is essential.

The XY selection function (also referred to as dimension order selection function) continuously chooses the output channel that belongs to the lowest dimension from all permissible free channels. For instance once applied to 2d mesh networks and if there exist free output channels both on x and y dimension, this selection function can come back the one on x direction. The XY selection function could be a fixed priority algorithm and its performance isn't very satisfying [25]. In [22] presented the implement of XY routing algorithm. it is deterministic routing algorithm that means this routing algorithm only provides a routing path for a pair of source and destination. More ever XY routing algorithm cannot avoid from deadlock appearance moreover as OE (odd- even) routing algorithm could be a distributed adaptive routing algorithm that is predicated on odd-even turn model. It exerts some restrictions, for avoiding and preventing from deadlock appearance. but, it is 2D dimensional routing.

Adaptive routing algorithm, named DyXY, along with an analytical model supported queuing theory for a 2D mesh has been presented in [23], though the authors claim that DyXY ensures deadlock-free and live lock-free routing. In [21] presented the algorithm, that is based on Dynamic XY (DyXY), is named enhance dynamic XY routing that the simulation results reveal that EDXY can have lower latency compared to those of alternative adaptive routing algorithms across all workload was examine, but they are not offerdeadlock free routing [18]. As solution for that [20], presented the deadlock free xyz routing algorithm which for 3D mesh NoC.In the adaptive XYZ routing [22], the main heading the fact and packets transverse depends on the position of present node and line extent of the buffer in x-direction and Y-direction as well as the response of the arbiter which will be directing the packets through the network. Adaptive routing algorithm needs more information about the network to bypass congested router in the mesh. These routing algorithms are obliquely more convoluted to apply therefore this may have more cost, more diameter and power consumption In this the performance of routing is not that much better than the xyz routing which are used in [10] this algorithm worked on torus based routing which overcome drawback of diameter and power consumption, but torus topology is not synthesizable.

IV. CONCLUSIONS

In this survey paper, we have elaborated into the four subsystems that are 3D Integration technology, 3D Network-on-Chip, and related topology and routing algorithm. Three dimensional (3D) Network-on-Chip (NoC) architecture combine the benefits of new integration technologies with NoC-style interconnection of large number of IP cores in a single chip as well 3D-NoC needs a huge number of vertical
links, the 3D Integration has a major limitation on the number of vertical interconnects using TSVs to be exploited. Additionally, 3D Integration enables the integration of differently fabricated with CMOS technologies, but constructing a homogeneous and regular network topology for such a heterogeneous system is very challenging. In literature survey in order to reduce the number of vertical links, scalability throughout, latency and for congestion aware 3D architecture the mesh and torus topology was used because of its spurious benefits. In Future we have to work on 3D NoC with mesh topology and XYZ routing algorithm in terms of diameter, latency and power consumption.

ACKNOWLEDGMENT

We acknowledge our senior faculty who has provided me their views in the selection of topic.

REFERENCES

[21] Zhuoyuan Li, Xianlong Hong,” Efficient Thermal via Planning Approach and Its Application in 3-D Floorplanning”, IEEE TRANSACTIONS ON COMPUTER-AIDED DESIGN OF INTEGRATED CIRCUITS AND SYSTEMS, VOL. 26, NO. 4, APRIL 2007
Detecting and Prevention of DDOS Attack in MANET using New Cracking Algorithm

Ms. Jasmine Patel1, Ms. Priya Wadhai2, Ms. Neha Agrawal3, Ms. Gitanjali Gaikwad4, Prof. Vaishnavi J. Deshmukh5

1,2 Student, Computer Engineering Department, D.B.N.C.O.E.T, Yavatmal
3 Lecturer, Computer Engineering Department, D.B.N.C.O.E.T, Yavatmal
4 Maharashtra, India.

Abstract—Security could be a weak link of network systems. The malicious usage and attacks have caused tremendous loss by impairing the functionalities of the pc networks. In an effort to reinforce security in MANETs several researchers have prompt and enforced new enhancements to the protocols and a few of them have prompt new protocols. Existing Manet routing protocols, like unintentional On-Demand Distance Vector Routing Protocol (AODV), don’t offer enough security defense capability. Distributed Denial of Service (DDoS) attack has become a significant downside to networks. During this paper, we have a tendency to introduce bottom-up approach, New Cracking algorithmic program, Prevention algorithmic program exploitation IDS node for sleuthing and dominant DDoS attack.

Keywords—Manet, Flooding attack, DDoS attack, Bottom Up, New Cracking algorithmic program, informatics Traceback.

II. INTRODUCTION

MANETs ar a sort of Wireless circumstantial network that typically includes a routable networking atmosphere on high of a Link Layer circumstantial network. A mobile circumstantial network (MANET) could be a self-configuring infrastructureless network of mobile devices connected by wireless, every device in an unengaged to move severally in any direction, and can so modification its links to different devicesofoftentimes. every should forward traffic unrelated to its own use, and so be a router. the first challenge in building a painter is arming every device to endlessly maintain the data needed to properly route traffic. Such networks might operate by themselves or could also be connected to the larger net. [20] Distributed denial-of-service attack (DDoS attack) is an effort to form a machine or network resource inaccessible to its supposed users. [19], painter could be a distributed system that includes wireless mobile nodes that may freely and dynamically self-organize into whimsical, temporary, and circumstantial network topologies, permitting seamless interconnections while not pre-existing communication infrastructure and central administration. Attributable it’s distinctive characteristics, painter is susceptible to numerous security threats, and it’s significantly at risk of the DDoS attack. [4]

Security Challenges & problems with Manets

MANETs use wireless media for transmission, that introduces security flaws to the networks, primarily anyone with the correct instrumentation and data of the present topology and also the protocols might acquire access to the network. each active and passive attacks like impersonation, eavesdrop-ping, message redirection, and traffic analysis, will be per-formed by associate degree active element. [10]

In specific eventualities, painter nodes could also be scattered over an outsized space. Some nodes or network elements could also be un-monitored or exhausting to watch, and exposed to the physical attacks. [2]

• as a result of MANETs don't have any central authority, this is often a serious barrier to security. the protection mechanisms utilized in wired networks, like Public Key Management, Node Authentication, and Determination of Node Behaviour, ar in reality terribly troublesome to attain with none central administration.

• circumstantial networks ar extremely dynamic in nature. Node joins and departures aren't predictable. Moreover, topology is often dynamic in circumstantial networks. [3].

III. MANET VULNERABILITY

Vulnerability could be a weakness in security system. a selected system is also susceptible to unauthorized information manipulation as a result of the system doesn't verify a user’s identity before permitting information access. painter is additional vulnerable than wired network. a number of the vulnerabilities ar as follows:-

Lack of centralized management: painter doesn’t have a centralized monitor server. The absence of management makes the detection of attacks troublesome as a result of it’s not east to observe the traffic during a extremely dynamic and enormous scale ad-hoc network.

Resource availability: Resource availability could be a major issue in painter. Providing secure communication in such ever-changing surroundings moreover as protection against specific threats and attacks, results in development of assorted security schemes and architectures. cooperative ad-hoc environments conjointly enable implementation of self-organized security mechanism.

Scalability: Attributable to quality of nodes, scale of ad-hoc network ever-changing all the time. Therefore measurability could be a major issue regarding security. Security mechanism ought to be capable of handling an oversized network moreover as tiny ones.

Cooperativeness: Routing algorithmic rule for MANETs typically assumes that nodes ar cooperative and non-malicious. As aresult a malicious wrongdoer will simply become a very important routing agent and disrupt network operation by disobeying the protocol specifications.
Dynamic topology: Dynamic topology and changeable nodes. Membership might disturb the trust relationship among nodes. The trust can also be disturbed if some nodes are detected as compromised. This dynamic behavior might be higher with distributed and reconciling security mechanisms. Limited power provide: The nodes in mobile ad-hoc network ought to take into account restricted power supply, which is able to cause several issues. A node in mobile ad-hoc network might behave during a stingy manner once it's finding that there's solely limited power provide.

Bandwidth constraint: Variable low capability links exists as compared to wireless network that ar additional liable to external noise, interference and signal attenuation effects. Adversary within the Network: The mobile nodes at intervals the painter will freely be part of and leave the network. The nodes within network can also behave maliciously. this can be arduous to notice that the behavior of the node is malicious. Thus this attack is additional dangerous than the external attack. These nodes ar known as compromised nodes.

No predefined Boundary: In mobile ad-hoc networks we tend to cannot exactly outline a physical boundary of the network. The nodes add a unsettled surroundings wherever they're allowed to hitch and leave the wireless network. As presently as associate adversary comes within the radio vary of a node it'll be able to communicate thereupon node. Eavesdropping impersonation; tempering, replay and Denial of Service (DoS) attack.[10].

Some specific DDoS sorts ar listed below:

**SYN Flooding:** The attack uses the weakness of the protocol handshaking. It sends associate abundance of protocol SYN packets to the victim. The victim opens lots of protocol connections and responds with ACK. SYN Flooding doesn't target specific OS, therefore it's going to attack any system supporting protocol.

**Ping of Death:** The aggressor sends the victim outsized information processing packets, that contain oversixty five,536 bytes. It may cause the victim machine to crash.[4]

**Smurf Attack:** The aggressor sends the printed address associate degree abundance of web managementMessage Protocol (ICMP) "echo-request" packets, that has the victim’s information processing because the supply address. The victim are going to be flooded with ICMP "echo-reply" packets [7].

**SSH method Table:** The aggressor overflows the SSH daemon within the victim system. it's kind of like the method table attacks.

**Protocol Reset:** The aggressor listens the traffic for the "tcpconnection" requests to the victim. Once such an invitation is found, the aggressor sends a spoofed protocol RESET packet to the victim and obliges it to prevent the protocol affiliation [9].

**Teardrop:** The aggressor creates a stream of information processing fragments with their offset field overlapped. The victim might crash once making an attempt to piece these misshapen fragments [8].

**UDP Packet Storm:** The aggressor spoofs a begin packet and builds a affiliation between 2 victim nodes, which provide a sort of UDP output services (such as "chargen" or "echo") to come up with varied traffic into the network [16].

**Smurf Attack:** The aggressor sends the printed address associate degree abundance of web management Message Protocol (ICMP) "echo-request" packets, that has the victim’s information processing because the supply address. The victim are going to be flooded with ICMP "echo-reply" packets . [7].

**IV. DDOS ATTACK IN MANETS**

Distributed denial of Service attacks sometimes happens in MANETS or in wireless networks. it's associate degree attack wherever multiple systems comprised along and target one system inflicting a denial of service (DoS). The target node is flooded with the info packets that system shutdowns, thereby denying service to legitimate users. The services under fire ar those of the “primary victim”, whereas the compromised systems accustomed launch the attack ar typically referred to as the “secondary victims.” [1]. Current MANets ar primarily at risk of 2 differing types of DDoS attacks:

- Active DDoS attack is associate degree attack once misbehaving node needs to bear some energy prices so asto perform the threat
- Passive DDoS attacks are chiefly owing to lack of cooperation with the aim of saving energy egotistically [14].

Nodes that perform active DDoS attacks with the aim of damaging alternative nodes by infecting network outage thought of as malicious whereas nodes that build passive DDoS attacks with the aim of saving battery life for his or her own communications are thought of to be self-serving [13] [1]. The attacks ar classified as:

**Modification Attack:** Modification may be a sort of attack once associate degree unauthorized party not solely gains access to however tampers with associate degree quality.

**Impersonation Attacks:** As there's no authentication of information packets in current adhoc network, a malicious node will launch several attacks in an exceedingly network by masquerading as another node i.e. spoofing. Fabrication Attacks: Fabrication is associate degree attack within which...
associate degree unauthorized party not solely gains the access however additionally inserts counterfeit objects into the system. [2].

V. ATTACK DETECTION WAYS IN MANET

Profile-based detection Profile-based detection is additionally referred to as behaviour-based detection. Profile-based detection defines a profile of traditional behaviour classifies any deviation of that profile as an anomaly. The belief of this kind of detection is that attacks area unit events distinguishable from traditional legitimate use of system resources. though this kind of anomaly observe area unit ready to detect novel attacks, they're prone to high false positive rate owing to the issue of clear segmentation between traditional and abnormal activities and also the use of lean or inadequate options to profile traditional behaviours. Specification-based detection Specification-based detection defines a group of constraints that describe the proper operation of a program or protocol and monitors the execution of the program with reference to the outlined constraints. it's been show that specification-based techniques live up to their promise of detection referred to as well as unknown attacks, whereas maintaining a really low rate of false positives. Since, the increasing quality of wireless networks to it of wired networks, security is being thought-about as a serious threat in them. Wireless network exposes a risk that Anunauthorized user will exploit and severely compromise the network. There may be completely different potential attacks in wireless network viz., active and passive attacks. therefore there's a desire for secured wireless system to research and observe variety of attacks.[1,17]

VI. REVIEW OF EXISTING DDoS DEFENCE TECHNIQUE

Bottom-up Detection and Prevention Techniques:

This detection theme is split into three phases:

Phase-I : Quality Reduction primarily based Attacks Once associate degree offensive host sends SYN(k) packet for brand new association to victim server victim server allocates memory for that host and sends SYN/ACK to it aggressor consumes one sequence range and waits to receive for ACK from offensive host. This state is named 0.5 open association state. additional and additional requests can accumulate and stock up the memory buffer at server aspect. aggressor send sizable amount SYN packets with spoofed supply information processing for preventing services to be granted to alternative legitimate requests. Therefore, no new request, together with legitimate requests, is processed and also the services of the system area unit disabled.

Phase-II Bottom-up Approach for discovery of protocol SYN Flood Attack The detection algorithmic rule desires the potential to detect any new beginning attacks not aboutthis happening ones. Also, if the extracted knowledge characteristics cannot match any signatures, those packets are going to be thought to be traditional network traffic and also the detection system ignores them. the entire detection method continues recursively until detection been terminated.

Phase-III hindrance Window-based management for traditional 0.5 Open Connection: during this approach we tend to planned a window limit per resource or per traffic combination. this permits U.S.A. to regulate however an explicit resource is consumed by a traffic category at any given time. once this limit is reached, incoming requests or packets seeking this resource area unit born or delayed at the QOS regulator till the server sends some quite indication that associate degree earlier request from this traffic category has freed its resources. once this happens, additional flows or requests is admitted. The windows limit quantifies the resource availability. TTL-based Packet Filtering Approach for Abnormal 0.5 Open Connection: Filtering all packets having an explicit TTL worth would end in the filtering of legitimate furthermore as attack packets. Hence, our TTL-based rate-limit theme includes rules for distinctive traditional from spoofed packets. It will this by observant protocol tripartite handshaking behaviours. throughout a standard 3 approach handshaking procedure, Syn(k), Ack(k + 1) + Syn(j) and Ack(j +1)is captured at the victim aspect.[4]

LPN DDoS Attack Mitigation ways—Negative Protection Node(LPNN) protects the victim node of a DDoS attack. The LPN node filters all the assaultive packages within the traffic whose destination is that the victim. additionally, the LPN acknowledges the supply informatics addresses adore the malicious traffic, An Attack Notification Message (ANM) is shipped to the victim node. The ANM includes the supply informatics addresses of concerned malicious attack agents. Then, the victim node broadcasts AN Attack data Message (AIM) packet towards the remote protection node (RPN). With the data in AIM, the RPN nodes filter off all the malicious packets at the supply facet. This mechanism aims to recover the service for destination protection node and to inform each alternative node to drop the RREQ from the malicious node. when doing this, the malicious nodes cannot transport traffic or build a route.[5]

Clustering Based Prevention Techniques – during this agglomeration technique the name and score worth of nodes to elect a cluster head and once a Cluster-Head is subjected to DDoS attack this is able to have manifolds consequences because the Cluster Heads type a virtual backbone and will be wont to maintain routing states data & route packets to nodes in their cluster. The design worked in 3 phases specifically as part I: name and Score based mostly Cluster Creation And Cluster Head choice part II: presents mentioning of a number of DDoS attacks like message bombing and cache poisoning, their detection ways part III: presents an impact frame packet format which may be used as a line of defence to regulate and mitigate from DDoS attacks over a name and score based mostly painter.[12]

Detection Algorithm using IDS node — during this rule foremost we tend to produce AN IDS node during which we tend to set AODV as a routing protocol. Then when the creation, our IDS node check the network configuration and capture sedimentation by finding that if any node is in its

ISSN: 2231-5381 http://www.ijettjournal.org Page 97
radio vary and conjointly ensuing hop isn't null, then capture all the data of nodes. Else nodes area unit out of vary or destination inaccessible. With the assistance of this data IDS node creates a standard profile that contains data like form of packet, in our case (protocol is AODV, pkt kind transmission control protocol, UDP, CBR), time of packet send and receive and threshold. hen making traditional profile and threshold checking is finished within the network i.e. if network load is smaller than or capable most limit and new profile is smaller than or capable most threshold and new profile is bigger than or capable minimum threshold then there's no any quite attack gift. Else there's AN attack within the network and realize the attack. For doing it compare traditional profile with every new trace worth i.e. check packet kind, count unknown packet kind, point in time of packet, sender of packet, receiver of packet. And when detection of any anomaly therein parameters then block that packet sender node. The projected mechanism eliminates the requirement for a centralized sure authority that isn't sensible in ADHOC network attributable to their self organizing nature. It protects the network through a self organized, absolutely distributed and localized procedure. It may also be applied for securing the network from alternative routing attacks by dynamic the safety parameters in accordance with the character of the attacks.[11]

New Cracking rule – during this rule a procedure is formed to tackle the continual issues occur within the net services. To avoid the continous logon to a selected data processor, this rule maintains a standing table, therein it keeps the informatics addresses of current users and their standing. If the actual informatics address has been signed on for a primary time, it makes the standing as real user. For 2, 3, four it marks as traditional user. For the fifth time it makes the actualinformatics address standing as aggressor. within the time calculations we tend to area unit solelycontemplate five times. User would like to server increase the time depends au courant the applying. After that, the user cannot permit get the service of that individual data processor. The service is denied theretospecific informatics address, the essential plan behind the projected system is to isolate ANd defend the netserver from vast volumes of DDoS request once an attack happens. conjointly a DDoS munition for shielding the net services is additionally projected. once a DDoS attack happens, the projected munition ensures that in an exceedingly network connected server data area unit managed while not corruption. This fresh designed system that effectively offers the provision of net services even throughout severe DDoS attacks. Our system is sensible and simply deployable as a result of it's clear to each net servers and shoppers and is absolutelycompatible with all existing network protocols.[15].

IP Traceback Algorithm – informatics traceback theme supported data metrics will effectively trace all attacks till their own LANs (zombies), last, our projected data metrics will considerably improve the performance of low-rate DDoS attacks detection and informatics traceback over the normal approaches. DDoS attacks detection metric is combined with informatics traceback rule and filtering technology along to create an efficient cooperative defence against network security threats in net. In hop-by-hop informatics tracing, the a lot of of hops the a lot of of tracing processes, so the longer time are going to be taken. so as to convenience forinformatics traceback rule analysis, we tend to classify 2 styles of traffic as native traffic and forward traffic. Thenative traffic of is that the traffic generated from its local area network, the forward traffic of is that the total of its native traffic and therefore the traffic forwarded from its immediate upstream routers. This data metrics will considerably improve the performance oflow-rate DDoS attacks detection and informatics traceback over the normal approaches.[16].

VII. Conclusions

There is associate degree horrific increase within the range of DDoS attack incidents. Not only, DDoS incidentsarea unit growing day by day however the technique to attack, botnet size, and attack traffic are attaining new heights. Effective defense measures required to stop and mitigate these attacks is that the current would like of the hour. In this paper, we introduce techniques for detection and dominant flooding and DDoS attacks inManet. they need most of the issues of wired networks and plenty of of a lot of owing to their specific features: dynamic topology, restricted resources, lack of central management points. First, we've got conferred specific vulnerabilities of this new surroundings. Then we've got surveyed the attacks that exploit these vulnerabilities and also the doable proactive and reactive solutions projected within the literature. Attacks area unit classified into passive and active attacks at the highest level. Then varied Preventive measures area unit mentioned so as to mitigate the results of DDOS attack in Manet. To conclude, Manet security may be a complicated and difficulttopic.

REFERENCES


http://en.wikipedia.org/wiki/Mobile_ad_hoc_network
“Audio Spotlighting System Using Ultrasonic Energy”

Monika Arkilwad, Aparna Rangari, Chetan Thote

Department of Electronics and Telecommunication Engineering,
Dr. Bhausaheb Nandurkar College of Engineering & Technology, Yavatmal

Abstract-

Audio spot lighting is a very recent technology that creates focused beams of sound similar to light beams coming out of a flashlight. By shining sound to one location, specific listeners can be targeted with sound without others nearby hearing it. It uses a combination of non-linear acoustics and some fancy mathematics. But it is real and is fine to knock the socks of any conventional loud speaker. This acoustic device comprises a speaker that fires inaudible ultrasound pulses with very small wavelength which act in a manner very similar to that of a narrow column. The ultra sound beam acts as an airborne speaker and as the beam moves through the air gradual distortion takes place in a predictable way due to the property of non-linearity of air. This gives rise to audible components that can be accurately predicted and precisely controlled. Joseph Pompei” holosonic Research Labs invented the Audio Spotlight that is made of a sound processor, an amplifier and the transducer. The American Technology Corporation developed the Hyper Sonic Sound-based Directed Audio Sound System. Both use ultrasound based solutions to beam sound into a focused beam. Audio spotlight can be either directed at a particular listener or to a point where it is reflected.

The targeted or directed audio technology is going to a huge commercial market in entertainment and consumer electronics and technology developers are scrambling to tap into the market. Being the most recent and dramatic change in the way we perceive sound since the invention of coil loud speaker, audio spot light technology can do many miracles in various fields like Private messaging system, Home theatre audio system, Navy and military applications, museum displays, ventriloquio systems etc. Thus audio spotlighting helps us to control where sound comes from and where it goes!

Keywords — Transducer, Ultrasonic amplifier, Ultrasonic Oscillator, Hypersonic sound technology.

I. INTRODUCTION

The Audio Spotlight & Hyper Sonic Sound Technology (developed by American Technology Corporation), uses ultrasonic energy to create extremely narrow beams of sound that behaves like beams of light. Audio spotlighting exploits the property of non-linearity of air. When inaudible ultrasound pulses are fired into the air, it spontaneously converts the inaudible ultrasound into audible sound tones, hence proved that as with water, sound propagation in air is just as non-linear, and can be calculated mathematically. A device known as a parametric array employs the non-linearity of the air to create audible by-products from inaudible ultrasound, resulting in an extremely directive, beamlike wide-band acoustical source. This source can be projected about an area much like a spotlight, and creates an actual specialized sound distant from the transducer. The ultrasonic column acts as an airborne speaker, and as the beam moves through the air, gradual distortion takes place in a predictable way.

This gives rise to audible components that can be accurately predicted and precisely controlled. The sound dispersion of the Audio Spotlight can be likened to a directional beam of light, like a flashlight. Its best use is for delivering sound to an isolated area. According to the supplied Manual, the most present sound is at a distance of around six feet while listening directly in front of the disc. At fifteen feet away from the disc, the sound level drops by about 9 decibels. Based on these specifications, its best use would be in a small environment where sound “bleed” is an issue. Museums utilize similar devices. In order to focus sound into a narrow beam, you need to maintain a low beam angle that is dictated by wavelength. The smaller the wavelength, the less the beam angle, and hence, the more focused the sound. Unfortunately, most of the human-audible sound is a mixture of signals with varying wavelengths—between 2cms to 17 meters (the human hearing ranges from a frequency of 20 Hz to 20,000 Hz). Hence, except for very low wavelengths, just about the entire audible spectrum tends to spread out at 360 degrees. To create a narrow sound beam, the aperture size of the source also matters—a large loudspeaker will focus sound over a smaller area. If the source loudspeaker can be made several times bigger than the wavelength of the sound transmitted, then a finely focused beam can be created. The problem here is that this is not a very practical solution. To ensure that the shortest audible wavelengths are focused into a beam, a loudspeaker about 10 meters across is required, and to guarantee that all the audible wavelengths are focused, even bigger loudspeakers are needed.

Here comes the acoustical device “AUDIO SPOTLIGHT” invented by Holosonics Labs founder Dr. F. Joseph Pompei (while a graduate student at MIT), who is the master brain behind the development of this technology.
II. LITERATURE REVIEW

This technology was originally developed by the US Navy and for under water in the mid-1960s, and was briefly investigated by Japanese researchers in the early 1980s, but these efforts were abandoned due to extremely poor sound quality (high distortion) and substantial system cost [1]. These problems were unsolved until a paper published by Dr. F. Joseph Pompei of the Massachusetts Institute of Technology in 1998 (105th AES Conv, Preprint 4853, 1998) fully described a working device that reduced audible distortion essentially to that of a traditional loudspeaker [2]. Since the early 1960s, researchers have been experimenting with creating directive low-frequency sound from nonlinear interaction of an aimed beam of ultrasound waves produced by a parametric array using heterodyning. Ultrasound has much shorter wavelengths than audible sound, so that it propagates in a much narrower beam than any normal loudspeaker system using audio frequencies known by the trademark name “Audio Spotlight”, a term first coined in 1983 by the Japanese researchers who abandoned the technology as infeasible in the mid-1980[1].

A transducer can be made to project a narrow beam of modulated ultrasound that is powerful enough, at 100 to 110 BSPL, to substantially change the speed of sound in the air that it passes through the air within the beam linearly and extracts the modulation signal from the ultrasound, resulting in sound that can be heard only along the path of the beam, or that appears to radiate from any surface that the beam strikes. This technology allows a beam of sound to be projected over a long distance to be heard only in a small well-defined area a listener outside the beam hears nothing. This effect cannot be achieved with conventional loudspeakers, because sound at audible frequencies cannot be focused into such a narrow beam.

Normal loudspeakers project a 90 degree or greater than 90 degree cone of sound. This is obviously much too wide for focused long distance auditory warning applications. One can at least take advantage of the proximity law that states that sound pressure level increases by 6 dB every time speaker to listener distance is halved. By locating the speaker as close as possible to the listener’s head, sound pressure level can be decreased, thereby decreasing the sound level in the spillover area. Even though it’s price is inexpensive, if auditory warning is applied for an application as shown in Figure 2, normal loudspeakers do not work well for this application.

The technique of using nonlinear interaction of high frequency wave to generate low frequency wave was originally pioneered by researcher developing underwater sonar technique in 1960s. In 1975, an article cited the nonlinear effect occurring in air. Over the next two decades, several large companies including Panasonic and Ricoh attempted to develop are proposed in [6]. A loudspeaker using this principle .They were successful in producing some sort of sound but with higher level of distortion (>50%). In 1990s Woody Norris a radar technician solved the parametric problem of this technology.

III. METHODOLOGY

The original low frequency sound wave such a human speech or music is applied into an audio spotlight emitter device. This low frequency signal is frequency modulated with ultrasonic ranging from 21KHz-28KHz. The output of the modulator will be the modulated from of original sound wave. Since ultrasonic frequency is used the wavelength of the combined signal will be in the order of few millimeters. Since the wavelength is smaller the beam angle will be around 3 degree, as a result the sound beam will be a narrow one with a small dispersion.

While the frequency modulated signal travels through the air, the non-linearity property of air comes into action which slightly changes the sound wave. If there is a change in a sound wave, new sounds are formed with in wave. Therefore if we know how the air affects the sound waves, we can predict exactly what new frequency will be added into the sound wave by the air itself. The new sound signal generated within the ultrasonic sound wave will be corresponding to the original information signal with a frequency in the range of 20-20KHz will be produced within the ultrasonic sound wave. Since we cannot here the ultrasonic sound waves we only hear the new sound that are formed by non-linear action of the air. Thus in an audio spotlighting there are no actual speakers that produce the sound but the ultrasonic envelope acts as the airborne speaker. The new sound produced virtually has no distortion of sound is freed from bulky enclosures. There are no woofers or crossovers. This technology is similar in that you can direct the ultrasonic emitter toward a hard surface, a wall for instance and the listener perceives the sound as coming from the spot on the wall. The listener does not perceive the sound as emanating from face of the transducer, but only from the reflection from the wall. For the maximum volume that trade show use demands, it is recommended that the audio spotlight speaker, more accurately called a transducer, is mounting no more than 3 meters from the avg. listener’s ears, or 5 meter in the air. The mounting hardware is constructed with a ball joint so that the audio spotlighting is easily aimed wherever the sound is desired.

![Fig1. Audio spotlighting Emitter](http://www.ijettjournal.org)
A. Ultrasound in Air

Researchers discovered that if short pulses of ultrasound were fired into water, the pulses were spontaneously converted into low frequency sound. Dr. Orhan Berktay established that water distorts ultrasound signals in a nonlinear, but predictable mathematical way. It was later found that similar phenomenon happens in air also. When inaudible ultrasonic sound pulses are fired into the air, the air spontaneously converted the inaudible ultrasound into audible sound tones, hence proving that as with water, sound propagation in air is just as non-linear, but can be calculated mathematically. As the beam moves through the air gradual distortion takes place giving rise to audible component that can be accurately predicted and precisely controlled.

The problem with firing off ultrasound pulses, and having them interfere to produce audible tones is that the audible component created are nowhere similar to the complex signals in speech and music which contains multiple varying frequency signals, which interfere to produce sound and distortion.

B. Berktay’s Equation

In 1965, Dr. H.O. Berktay published the first accurate and more complete theory of distortion of ultrasound signal in air. He uses the concept of modulation envelope. The air demodulates the modulated signal and the demodulated signal depends on the envelope function. Berktay assumes the primary wave has the form

\[ P_1(t) = P_1 E(t) \sin(Wct) \]

Where we is the carrier frequency and \( E(t) \) is the envelope function which in this case is the speech or music signal

The secondary wave or demodulated wave is given by

\[ P_2(t) = \frac{d}{dt} E(t) \]

This is called berktay’s far field solution. The berktay’s solution states that the demodulated signal is proportional to the second time derivative of the envelope squared. This is the fundamental expression for the output resulting from the distortion due to air.

IV. BLOCK DIAGRAM

Fig3.Block diagram of audio spot lighting

A. Components of spotlighting system

1. Power supply
2. Frequency oscillator
3. Modulator
4. Audio signal processor
5. Microcontroller
6. Ultrasonic amplifier
7. Transducer

1. Power supply: - Like all electronics system, the audio spotlighting system works off DC voltage. Ultrasonic amplifier requires 48V DC supply for its working and low voltage for microcontroller unit and other process management

2. Frequency oscillator: - The frequency oscillator generate ultrasonic frequency signal in the range which is required for the modulation of information signal.

3. Modulator: - In order to convert source signal material into ultrasonic signal modulator scheme is required which is achieve through modulator. In addition, error correction is needed to reduce distortion without loss of efficiency by using DSB Modulator. The modulation index can be reduced to decrease distortion.

4. Audio signal processor: - The audio signal sent to electronic signal processor circuit where equalization and distortion control are performed in order to produce a good quality sound signal.

5. Microcontroller: - A dedicated microcontroller circuit takes care of functional management of in future version. It is expected that the whole process like functional management signal processing double sideband modulation and even switch mode power supply would be effectively taken care of by single embedded IC.

6. Ultrasonic amplifier: - High efficiency ultrasonic power amplifier amplifies the frequency modulated wave in order to match the impedance of the integrated transducers. So that the output of the emitter will be more powerful and can cover more distance.

7. Transducer: - It is 1.27cm thick and 17” in diameter. It is capable of producing audibility up to 200 meters with better clarity of sound. It has the ability of real time sound reproduction with “0” lag. It can be wall overhead or flush mounted. This transducer are arranged in the form of an array called parametric array in order to propagate the ultrasonic signal from the emitter and thereby to exploit non-linearity Property of air amplifier requires 48V DC supply for
its working and low voltage for microcontroller unit and other process management.

V. TYPES OF AUDIO SPOTLIGHTING

A. Directed Audio and Projected Audio

There are two ways to use Audio Spotlight. First, it can direct sound at a specific target, creating a contained area of listening space which is called “Direct Audio”. Second, it can bounce off of a second object, creating an audio image. This audio image gives the illusion of a loudspeaker, which the listener perceives as the source of sound, which is called “projected Audio”. This is similar to the way light bounces off of objects. In either case, the sound’s source is not the physical device you see, but the invisible ultrasound beam that generates it.

Hyper Sonic Sound technology provides linear frequency response with virtually none of the forms of distortion associated with conventional speakers. Physical size no longer defines fidelity. The faithful reproduction of sound is freed from bulky enclosures. There are no, woofers, tweeters, crossovers, or bulky enclosures. Thus it helps to visualize the traditional loudspeaker as a light bulb, and HSS technology as a spotlight, that is you can direct the ultrasonic emitter toward a hard surface, a wall for instance, and the listener perceives the sound as coming from the spot on the wall. The listener does not perceive the sound as emanating from the face of the transducer, only from the reflection off the wall. Contouring the face of the HSS ultrasonic emitter can tightly control Dispersion of the audio wave front. For example, a very narrow wave front might be developed for use on the two sides of a computer screen while a home theater system might require a broader wave front to envelop multiple listeners.

VI. APPLICATIONS

1. Automobiles: Beam alert signal can be directly propagated from an announcement device in the dashboard to the driver. Presently Mercedes Benz buses are fitted with audio spotlighting speaker so that individual travelers can enjoy the music.

2. Retail sales: Provide targeted advertising directly at the point of purchase.

3. Safety officials: Portable audio spotlighting device for communication with a specific person in a crowd of people.

4. Public announcement: Highly focused announcement in noisy environment such as subways, airport, traffic intersections etc.

5. Entertainment system: In home theatre system tear speaker can be eliminated by the implementation of audio spotlighting and the properties of sound can be improved.

6. Museums: In museums audio spotlighting can be used to describe about a particular object to a person standing in front it, so that the order person standing in front of another object will not be able to here the description.

7. Military applications: Ship to ship communication and shipboard announcements.

VII. ADVANTAGES

1. Can focus sound only at the place you want.

2. Ultrasonic emitter device are thin and flat and do not require a mounting cabinet.

3. The focused or directed sound travels much faster in a straight line than conventional loudspeaker.

4. Dispersion can be controlled very narrow or wider to cover more listening area.

5. Can reduce or eliminate the feedback from microphone.

6. Highly cost effective as the maintenance required is less as compared to conventional loud speakers and have longer life span.

7. Requires only same power as required for regular speakers.

8. There is no lag in reproducing the sound.

VIII. CONCLUSION

“Being the most radical technological development in acoustics since the coil loudspeaker was invented in 1925. The audio spotlight will force people to rethink their relationship with sound.”

So we can conclude- Audio Spotlighting really “put sound where you want it” and will be “A REAL BOON TO THE FUTURE.”
ACKNOWLEDGMENT

We avail this opportunity to express our deep sense of gratitude and whole hearted thanks to Head of the Department, Electronics and Tele-communication Engineering, Prof. S. A. Chavan for their invaluable guidance and encouragement to embark this paper.

We are thankful to whose esteemed suggestions and encouragement from time to time have always been unparalleled stimuli for us to travel eventually towards completion of this seminar.

We also put sincere thanks to Dr. V. G. Arajpure (Principal DBNCOET Yavatmal) for constant motivation and providing necessary infrastructure

IX. REFFERENCE


An Overview of Audio Steganography Techniques Using Phase Encoding and LSB

Omkumar G. Paratkar¹, Aniket N. Kale²

¹ Student, Computer Engineering, SGBAU Amravati University, India
² Student, Computer Engineering, SGBAU Amravati University, India

Abstract—In this paper there is study of different techniques of audio steganography exploitation totally different algorithms like LSB approach and Phase Coding. We've tried to cover some approaches that help in audio steganography. As we know that it is the art and science of writing hidden messages in such a way that nobody, except the sender and intended recipient, suspects the existence of the message, a form of security through obscurity. In steganography, the message used to hide secret message is termed host message or cover message. Once the contents of the host message or cover message are changed, the resultant message is thought as stego message. In different words, stego message is combination of host message and secret message. Audio steganography needs a text or audio secret message to be embedded inside a cover audio message. Due to availability of redundancy, the cover audio message before steganography, stego message once steganography remains same for data hiding.

Keywords- Audio Data Hiding, Phase Coding, LSB, HAS

1. INTRODUCTION

The past few years have seen an explosion within the use of digital media. trade is creating important investments to deliver digital audio, image, and video data to consumers and customers. a brand new infrastructure of digital audio, image, and video recorders and players, on-line services, and electronic commerce is apace being deployed. At constant time, major companies are changing their audio, image, and video archives to an electronic kind. Digital media provide many distinct benefits over analog media: the standard of digital audio, image, and video signals is beyond that of their analog counterparts. Editing is simple as a result of one will access the precise distinct locations that ought to be modified. Repetition is easy with no loss of fidelity. A replica of a digital media is identical to the initial. Digital audio, image, and videos are simply transmitted over networked data systems. These benefits have opened several new possibilities. In specific, it's attainable to cover information (information) at intervals digital audio, image, and video files. The knowledge is hidden within the sense that it's perceptually and statistically undetectable. With several schemes, the hidden data can still be recovered if the host signal is compressed, edited, or regenerate from digital to analog format and back. Data embedding additionally provides a mechanism for embedding important management, descriptive, or reference data in a given signal.

For embedding important management, descriptive, or reference data in a given signal. To encode secret messages in audio is that the most difficult technique to use once managing Steganography. This can be as a result of the human auditory system (HAS) has such a dynamic range that it will listen over. to place this in perspective, the (HAS) perceives over a range of power larger than one thousand thousand to at least one and a spread of frequencies larger than one thousand to at least one creating it extraordinarily exhausting to feature or take away information from the original data structure. The only weakness within the (HAS) comes at attempting to differentiate sounds (loud sounds drown out quiet sounds) and this can be what encode be exploited to write secret messages in audio without being detected. Audio masking is the effect by which a faint but audible sound becomes inaudible in the presence of another louder audible sound, i.e. the masker.

2. METHODS FOR HIDING INFORMATION:

There are various types of the methods that are used to hide the information within any Digital media that may be image, Audio and Video files etc. The two methods to hide the information within the digital media(Image, Audio, Video file) are Phase coding and LSB respectively.

2.1 PHASE CODING:

Phase coding substitutes the phase of an initial audioSegment with a reference phase that represents the Hidden information. this may be thought of, as kind of an encryption for the audio signal by victimization what's known as discrete Fourier transform (DFT), which is nothing over a transformation algorithmic rule for the audio signal. Human auditory system (HAS) can’t acknowledge the phase change in audio signal as simple it will acknowledge noise within the signal. The phase coding methodology exploits this fact. this method encodes the secret message bits as phase shifts within the phase spectrum of a digital signal, achieving an inaudible encoding in terms of signal-to-noise ratio.
Phase coding addresses the disadvantages of noise-inducing strategies of audio steganography. Phase coding depends on the actual fact that the phase components of sound don't seem to be as perceptible to the human ear as noise is. Instead of introducing perturbations, the technique encodes the message bits as phase shifts within the section spectrum of a digital signal, achieving an inaudible encoding in terms of signal-to-perceived noise ratio. Phase committal to writing is explained within the following procedure:

- The original sound signal is broken up into smaller segments whose lengths equal the size of the message to be encoded.
- A discrete Fourier transform (DFT) is applied to every segment to form a matrix of the phases and Fourier transform magnitudes.
- Phase variations between adjacent segments square measure calculated.
- Phase shift between consecutive segments are detected. In alternative words, the absolute phases of the segments are often changed however the relative phase variations between adjacent segments should be preserved. Thus the secret message is only inserted within the phase vector of the primary signal segment as follows:

\[
\text{phase}_{\text{new}} = \begin{cases} 
\pi/2 & \text{if message bit = 0} \\
-\pi/2 & \text{if message bit = 1} 
\end{cases}
\]

- A new phase matrix is created. A discrete Fourier transform (DFT) is applied to every segment to form a matrix of the phases and Fourier transform magnitudes.
- Victimization the new phase matrix and original magnitude matrix, the sound signal is reconstructed by applying the inverse DFT and then concatenating the sound segments back together.

To extract the secret message from the sound file, the receiver should understand the segment length. The receiver can then use the DFT to induce the phases and extract the information. One disadvantage related to part secret writing could be a low information transmission rate owing to the actual fact that the secret message is encoded within the initial signal segment only. This may well be addressed by increasing the length of the signal phase. However, this would change phase relations between every frequency component of the segment additional drastically, making the encoding easier to notice. As a result, the phase coding technique is employed when solely atiny low quantity of data, like a watermark, must be concealed.

### 2.2 LEAST SIGNIFICANT BIT

Least significant bit (LSB) coding is that the easiest method to introduce into a very digital audio file. By substituting the least important significant every sampling purpose with a binary message, LSB coding permits for a large quantity of information to be encoded. Among many various information concealing techniques projected to introduce secret message at intervals audio file, the LSB information concealing technique is one in every of the only ways for inserting data into digital signals in noise-free environments, that simply embeds secret message-bits in a very set of the LSB planes of the audio stream.

This projected system is to supply an honest, efficient method for concealment the information from hackers and sent to the destination during a safe manner. This project system won’t modification the scale of the file even once encoding and conjointly appropriate for any style of audio file format. Encryption and secret writing techniques are reused to build the safety system sturdy. Low-bit encoding embeds secret information into the smallest amount vital bit (LSB) of the audio file. The data rate is 1KB per second per rate (44 kbps for a forty four kilocycle per second sampled sequence). This technique is simple to incorporate.

### 3. IMPLEMENTATION:

Now a days, we have heard number of times a term hacking. A hacking in nothing but unauthorized access of the data at the time of data transmission. We need to prevent our data from this kind of access, to overcome this problem we implement some new logic through which we can damage to the hacker’s system or restart system also send the acknowledgment to both the side of transmission to know their data get hacked. Thus from this logic we may get the better robustness and security for the communication.

### 4. LITERATURE REVIEW:

In an audio steganography system, secret messages are embedded with digital sound. The secret message is embedded by slightly altering the binary sequence of a sound file. Existing audio steganography software can
embed messages in the various audio files such as WAV, even MP3 sound files. Embedding secret messages in digital sound is usually a more difficult process than embedding messages in other media, such as digital images. This is because the human auditory system (HAS) has such a dynamic range that it can listen over. In order to conceal secret messages successfully, a variety of methods for embedding information in digital audio have been introduced. These methods range from rather simple algorithms that insert information in the similar to watermarks on actual paper and are sometimes used as digital watermarks. Masking images entails changing the luminance of the masked area. The smaller the luminance change, the less of a chance that it can be detected.

5. RESULT

In this paper we have introduced a method i.e. phase coding and LSB for Audio steganography. The audio steganography is provide a good, efficient method for hiding the data from hackers and sent to the destination in a safe manner. The audio file may contain the digital media. Audio steganography is also ideally suited for covert communications. The steganography can securely hide large amounts of potentially encrypted information in audio, image, and video data. Encoding secret messages in audio is the most challenging technique to use when dealing with Steganography. A perfect audio steganographic technique aim at embedding data in an imperceptible, robust and secure way and then extracting it by authorized people. Hence, up to date the main challenge in digital audio steganography is to obtain robust high capacity steganographic systems. Leaning towards designing a system that ensures high capacity or robustness and security of embedded data has led to great diversity in the existing steganographic techniques.

6. REFERENCES:


Abstract
This paper focuses on the lossless image compression without any duplication problem by using duplication free run length coding. When an image is stored in raw as an array of pixels, the file size is usually large and changes proportionally with the resolution and number of color channels. For that, an image is usually compressed for storage as well as transmission purposes. Depending on the application, an image can be compressed so that a perfect reconstruction is possible (i.e. lossless) or only an approximate of the original image can be regenerated (i.e. lossy). The existing method in first RLC algorithm (Traditional RLC) is to encode two consecutive pixels of same intensity into a single code word hence gaining compression. However the traditional RLC method suffers from duplication problem. An entropy rule based generative coding method is used to generate variable length code words and these code words are assigned to pixel intensity values based on their probability of occurrence and does not require flag bits for differentiation between intensities and runs which is the case of TRLC and hence overcomes duplication problem.

1. Introduction
1.1 IMAGE
Image is a collection of pixels which area arranged in regular manner to form rectangular patterns. Basically image is a two dimensional light intensity function f(x, y), where x, y are spatial coordinates and f(x, y) is a grey value or intensity. An image is essentially a 2-D signal processed by the human visual system. The signals representing images are usually in analog form. However, for processing, storage and transmission by computer applications, they are converted from analog to digital form. A digital image is basically a 2-Dimensional array of pixels. Images form the significant part of data, particularly in remote sensing, biomedical and video conferencing applications. The use of and dependence on information and computers continue to grow, so too does our need for efficient ways of storing and transmitting large amounts of data. There are two types of images
1) Digital image
2) Binary image

1.2 DIGITAL IMAGE
An image is said to be digital image if the amplitude of the image and spatial coordinates are discrete quantities. A digital image is composed by finite number of elements called image elements or pixel elements or pixels

1.3 BINARY IMAGE
An image is said to a binary image if the amplitude value is either one or zero i.e; binary image consists of only black and white colours. Binary image can be represented by a set of ordered triplet.

2. Literature Review
The history of compression begins in the 1960s. An analogue videophone system had been tried out in the 1960s, but it required a wide bandwidth and the postcard-size black-
and-white pictures produced did not add appreciably to voice communication!

There are two closely interlinked reasons for compression.
1) Firstly, to reduce the size of stored or transmitted files to manageable sizes, or to reduce the time it would take to transmit these files to another computer.

2) Secondly, to reduce the real-time bandwidth required to transmit time-sensitive video data across a communication link.

Communications equipment like modems, bridges, and routers use compression schemes to improve throughput over standard phone lines or leased lines. Compression is also used to compress voice telephone calls transmitted over leased lines so that more calls can be placed on those lines.

Basic types of compression:
Compression comes in two basic flavors: lossless (or information preserving) and lossy.

- **Lossless compression** with lossless compression, data is compressed without any loss of data. It assumes you want to get everything back that you put in i.e., we can reconstruct a perfect reproduction of the original from the compression. Critical financial data files are examples where lossless compression is required.

- **Lossy compression** With lossy compression, it is assumed that some loss of information is acceptable. When you reconstruct the information from the compressed data, you get something close to but not exactly the same as the original. You can get more compression by allowing the algorithm to lose more information. Lossy compression algorithms usually have an adjustable parameter to control this compression vs. quality tradeoff.

3. Methodology

**Duplication Free RLC**
In this section, we propose a lossless image compression algorithm based on the aforementioned EBD and EBI codeword's. The purpose of not defining y in EBI codeword’s is to make the codeword’s adaptive to the requirements of a particular encoding specification, if any. For that, permutations are assigned to y according to the encoding requirements during the actual encoding stage. Here, the proposed code is applied to implement RLC and to uniquely differentiate flags in an efficient way. The encoding stage starts by assigning codeword’s to the intensity levels. The assignment is done according to the probability of occurrence (PoO) of each level such that the intensity levels of higher PoO are assigned to shorter codeword’s, and vice versa. The pixels are then scanned in raster order and encoded by using EBI codeword’s. Since the ending group y is not defined at the pre-encoding stage, y in each codeword is utilized as a flag to determine the status of the next (neighbor) codeword. In particular, the flag can be set to indicate three status, namely, S1, S2 and S3, where S1 < S2 < S3. Each status is a permutation that satisfies R5. In case the encoder reads a run of three or more pixels of the same intensity value, these pixels are encoded as a codeword that represents their intensity level. y in this codeword is set to S3 to indicate that the next codeword encodes the count of pixels in the run. Here, the count of run is encoded by using the EBD codeword’s described, which can encode an infinite number of pixels when considering more than one general expression. In case the encoder reads a run of two pixels, these pixels are encoded into a single codeword in which its y is set to S2, indicating that the encoded pixel intensity value is repeated twice and the next codeword encodes the intensity level of the
next pixel(s). To the best of our knowledge, DF-RLC is the first RLC approach that gains a reduction (in terms of file size) from a run of only two pixels by encoding them to a single codeword. Other RLC approaches encode the pair of pixels to two codeword’s, i.e., one codeword for the intensity value and another codeword for the count. Some approaches just leave the pair of pixels as they are. Finally, in case the encoder reads a single pixel (i.e., not repeated or run is unity), this pixel is encoded by a single codeword, and its y is set to S1 to indicate that the next codeword is the intensity level of the next pixel(s). An image is thus encoded by using combinations of the aforementioned encoding modes. The fore mentioned encoding modes. It is obvious that the proposed coding method overcomes the duplication problem and hence we name the proposed method as duplication free run-length coding (DF-RLC hereinafter.

EXAMPLE:

- Consider the sequence 22222555533336799 and by using duplication free RLC
- [2 5 3 4 6 1 7 1 9 2]
- Here the runs of 2, 3 are 5 and 4 i.e. >3 and therefore 5 and 4 will be replaced by $s_4$
- The runs of 6 and 7 are 1, 1 therefore it will be replaced by $s_1$, the run of 9 is 2, therefore it will be replaced by $s_2$.
- [2 $s_4$ 3 $s_4$ 6 $s_1$ 7 $s_1$ 9 $s_2$]
- Where $s_4=\{5, 4\}$, $s_1=1$, $s_2=10$.
- The number of bits transmitted are $5*8+2*8+1+2=59$

4. Applications
DFRLC can be applied in
- medical imaging
- satellite imaging
- artistic computer graphic design

- Tele video conferencing
- Remote sensing

5. Results
DF-RLC is implemented in MATLAB and tested with different grey scale images. The compression ratios are calculated using the three techniques i.e. RLC, TRLC, DF-RLC and the results are tabulated. It is observed that all compression ratios of the proposed DF-RLC are higher than those of TRLC1 and TRLC2, except for images that are of high spatial activity (i.e., ‘Lena’, ‘numbers’, ‘boat’ and ‘house’) in which case their file sizes are increased due to DF-RLC encoding. The increase in file size is due to the assignment of longer codeword’s to pixel intensity levels that rarely but inevitably occur.

DFRLC is applied to images like cameraman, tank and Lena. It is observed that compression ratio for cameraman and Lena images are not improved using DF-RLC. This is because they have high spatial activity. For tank image compression ratio is improved as it is having low spatial activity.

Photo 5.1 CAMERA MAN IMAGE

| Table I Compression Ratios of cameraman Image Using RLC, TRLC, DFRLC |
|---------------------------------|-----------------|-----------------|
| Cameraman image                | Compressi       | Saving          |
|                                 | on ratio        | percentage      |
| RLC                             | 1.1702          | 14.5%           |
| TRADITIONAL RLC                 | 1.2832          | 22.06%          |
| DUPLICATION FREE RLC            | 0.9273          | 7.83%           |
TABLE II Compression Ratios of Lena Image Using RLC, TRLC, DFRLC

<table>
<thead>
<tr>
<th>Lena image</th>
<th>Compression ratio</th>
<th>Saving percentage</th>
</tr>
</thead>
<tbody>
<tr>
<td>RLC</td>
<td>1.4211</td>
<td>29.63%</td>
</tr>
<tr>
<td>TRADITIONAL RLC</td>
<td>5.2896</td>
<td>81.09%</td>
</tr>
<tr>
<td>DUPLICATION FREE RLC</td>
<td>1.0424</td>
<td>4.06%</td>
</tr>
</tbody>
</table>

6.Conclusions

An entropy rule based generative coding method was proposed to produce codeword that are capable of encoding both intensity level and different flag values into a single codeword. The proposed method gains compression by reducing a run of two pixels to only one codeword. Our method has no duplication problem that traditional RLC usually suffer and the number of occurrences that can be encoded by a single run is infinite. Experimental results verified that duplication free RLC is less efficient in case of images with high spatial activity. The extension of the proposed method is to improve the compression ratio by using statistical consideration.

7.Future Scope

The above proposed method is called GLOBAL scenario. It improves compression ratio of images having low spatial activity. Hence BLOCKING scenario is used as an extension of it i.e. the whole image is divided into different blocks of different sizes.

References


Priyanka S. Burhan
Place of birth- Rajura, MH, DOB:- 05\textsuperscript{th} September 1991. B.E-B.M.E, Dr Bhausaheb Nandurkar College of Engg & Tech, Yavatmal,M.E-B.M.E(Pursuing), Bharatratna Indira Gandhi College of Engg, Solapur.

Pujari Aishwarya R.
A 3GPP LTE-Advance Turbo Encoder and Turbo Decoder

Manish G. Shingnare¹, Rini D. Kedarpawar², Piyush D. Hegu³

¹²B.E. Final year, E&TC Department, SGB Amravati University, DBNCOET Yavatmal, M.S. (India)
³Assistant Professor, E&TC Department, SGB Amravati University, DBNCOET Yavatmal, M.S. (India)

Abstract—This paper presents the designing of a 3GPP LTE Advanced Turbo Encoder and Decoder by using the convolutional interleaver. The high-throughput 3GPP Advance Turbo code requires Encoder and Decoder architecture. Interleaver is known to be the main obstacle to this implementation and introduces latency due to the collisions introduced in accesses to memory. In this paper, we propose a low-complexity Soft Input Soft Output (SISO) Turbo Encoder and Decoder for memory architecture to enable the turbo coding that achieves minimum latency. Design trade-offs in terms of area and throughput efficiency are explored to find the optimal architecture. The proposed Turbo Encoder and Decoder has been modelled using Simulink; various test cases are used to estimate the performances. The results are analysed and achieved 50% reduction in computation time along with reduced BER (e−3). The hardware of the Turbo Encoder and Turbo Decoder has been modelled in Verilog HDL, simulated in Modelsim, synthesized using TSMC 65 nm Synopsys Design compiler.

Keywords: Convolutional Interleaver, Turbo decoder, VLSI, 3GPP LTE.

VIII. INTRODUCTION

A 3GPP Long Term Evolution (LTE) is a set of enhancements to the 3G Universal Mobile Telecommunications System (UMTS), has received tremendous attention recently and is considered to be a very promising 4G wireless technology. For example, Verizon Wireless has decided to deploy LTE in their next generation 4G evolution. One of the main advantages of 3GPP LTE is high throughput. For example, it provides a peak data rate of 326.4 Mbps for a 4×4 antenna system, and 172.8 Mbps for a 2×2 antenna system for every 20 MHz of spectrum. Furthermore, LTE-Advance, the further evolution of LTE, promises to provide up to 1 Gbps peak data rate. The channel coding scheme for LTE is Turbo coding. The Turbo decoder is typically one of the major blocks in a LTE wireless receiver. Turbo decoders suffer from high decoding latency due to the iterative decoding process, the forward–backward recursion in the maximum a posteriori (MAP) decoding algorithm and the interleaving/de-interleaving between iterations. Generally, the task of an interleaver is to permute the soft values generated by the MAP decoder and write them into random or pseudo-random positions.

IX. LITERATURE REVIEW

Table 1 summarizes the measurement results of the turbo decoder and provides a comparison to other published LTE turbo decoders. At the maximum clock frequency of 410MHz, the highest block size of 6144 is processed with a throughput of 1013Mb/s at a code rate of 0.95 using a window size of 30 at 5.5 iterations. At low code rates, optimum decoder performance can be achieved with a window size of only 14, and the throughput additionally boosted to 1161Mb/s with overlapping half-iterations. At maximum throughput the power consumption is 966mW, which corresponds to an unprecedented energy efficiency of 0.17nJ/bit/iter. In strong contrast to the other state-of-the-art implementations in TABLE 1, our decoder has been optimized for the entire range of code rates, which enables to maintain high throughput even for highest code rates without any concessions in BER performance.
III. LTE ENABLING TECHNOLOGIES

LTE aims at better spectral flexibility, higher data rates, low latency, improved coverage and better battery lifetime. Table 2 lists the key targets of LTE. To achieve the targets, LTE employs the enabling technologies: Orthogonal Frequency Division Multiple Access (OFDMA), Single Carrier Frequency Division Multiple Access (SC-FDMA) and Multiple Input Multiple Output (MIMO). LTE employs OFDMA for downlink and SC-FDMA for uplink datatransmissions.

Table 2. LTE Performance Requirements

<table>
<thead>
<tr>
<th>Metric</th>
<th>Requirements</th>
</tr>
</thead>
<tbody>
<tr>
<td>Latency</td>
<td>1. Control-plane: Less than 100 msec to establish U-plane</td>
</tr>
<tr>
<td></td>
<td>2. User-plane: Less than 10 msec from UE to Server</td>
</tr>
<tr>
<td>Mobility</td>
<td>1. Optimized for low speeds (0-15 km/hr)</td>
</tr>
<tr>
<td></td>
<td>2. High performance at speeds up to 120 km/hr</td>
</tr>
<tr>
<td></td>
<td>3. Maintain link at speeds up to 350 km/hr</td>
</tr>
<tr>
<td>Coverage</td>
<td>1. Full performance up to 5 km</td>
</tr>
<tr>
<td></td>
<td>2. Slight degradation 5 km – 30 km</td>
</tr>
<tr>
<td></td>
<td>3. Operation up to 10km should not be precluded by standard</td>
</tr>
<tr>
<td>Spectral Flexibility</td>
<td>1.4, 3, 5, 10, 15 and 20 MHz</td>
</tr>
<tr>
<td>Peak data rate</td>
<td>1. Downlink (2 Ch MIMO): 100 Mbps</td>
</tr>
<tr>
<td></td>
<td>2. Uplink (Single Ch Tx): 50 Mbps (20 MHz ch)</td>
</tr>
<tr>
<td>Supported antenna configurations</td>
<td>1. Downlink: 4x2, 2x2, 1x2, 1x1</td>
</tr>
<tr>
<td></td>
<td>2. Uplink: 1x2, 1x1</td>
</tr>
<tr>
<td>Spectrum efficiency</td>
<td>1. Downlink: 3 to 4 times HSDPA Rel. 6</td>
</tr>
<tr>
<td></td>
<td>2. Uplink: 2 to 3 times HSDPA Rel. 6</td>
</tr>
</tbody>
</table>

IV. FUNDAMENTALS OF TURBO CODES

In order to explain the proposed Turbo decoder architecture, the fundamentals of Turbo codes are briefly described in this section.

A) Turbo Encoder Structure:

As shown in Fig.1, the Turbo encoding scheme in the LTE standard is a parallel concatenated convolutional code with two 8-state constituent encoders and one convolutional interleaver. The function of the convolutional interleaver is to take a block of N-bit data and produce a permutation of the input data block. From the coding theory perspective, the performance of a Turbo code depends critically on the interleaver structure. The basic LTE Turbo coding rate is 1/3. It encodes an N-bit information data block in to a code word with 3N+12 data bits, where 12 tail bits are used for trellis termination. The initial value of the shift registers of the 8-state constituent encoders shall be all zeros when starting to encode the input information bits. LTE has defined 188 different block sizes.

The convolutional encoder can be represented as follows:

- $g_0 = I + D + D^2 + D^3 + D^6$
- $g_1 = I + D^2 + D^3 + D^5 + D^6$

The convolutional encoder basically multiplies the generator Polynomials by the input bit string, as follows [6]:

- $A(x) = g_0(x) * I(x) = a \ b \ c \ ... \ g$
- $B(x) = g_1(x) * I(x) = P \ Q \ R \ ... \ V$

Interleaving the two outputs from the convolutional encoder yields

$$E(x) = aPbQcR \ ... \ gV,$$

Which can also be written as:

$$E(x) = (a0 \ b0 \ c0 \ ... \ g0) + (0P0Q0R \ ... \ 0V) = A(x2) + x*B(x2)$$

Therefore,

$$E(x) = A(x2) + x*B(x2)$$ and $$A(x2) = g0(x2) + I(x2)$$

And

$$B(x2) = g1(x2) * I(x2),$$

With the following:

$$E(x) = g0(x2) * I(x2) + x * gI(x2) * I(x2)$$
\[ I(x_2) = (g_0(x_2) + x \cdot g_1(x_2)) \]
\[ I(x_2) = G(x) \]

Where,
\[ G(x) = g_0(x_2) + x \cdot g_1(x_2) \]

i.e. \((x) = 1 + x + x^2 + x^4 + x^5 + x^6 + x^7 + x^{11} + x^{12} + x^{13}\)

Fig. 1. Structure of rate 1/3 Turbo encoder in LTE

B) Turbo Decoder Structure:

The basic structure of a Turbo decoder is functionally illustrated in Fig.2. A turbo decoder consists of two maximum a posteriori (MAP) decoders separated by an interleaver that permutes the input sequence. The decoding is an iterative process in which the so-called extrinsic information is exchanged between MAP decoders. Each Turbo iteration is divided into two half iterations. During the first half iteration, MAP decoder 1 is enabled. It receives the soft channel information (soft value \(L_s\) for the systematic bit and soft value \(L_{p1}\) for the parity bit) and the a priori information \(L_a1\) from the other constituent MAP decoder through deinterleaving to generate the extrinsic information \(L_{e1}\) at its output. Likewise, during the second half iteration, MAP decoder 2 is enabled, and it receives the soft channel information (soft value \(L_s\) for a permuted version of the systematic bit and soft value \(L_{p2}\) for the parity bit) and the a priori information \(L_a2\) from MAP decoder 1 through interleaving to generate the extrinsic information \(L_{e2}\) at its output. This iterative process repeats until the decoding has converged or the maximum number of iterations has been reached.

Fig.2 (b): Basic structure of an iterative Turbo decoder [7].
(1) Iterative decoding based on MAP decoders.
(2) Forward/backward recursions on the trellis diagram.

C) Convolutional Interleaver:

A convolutional interleaver consists of \(N\) rows of shift registers, with different delay in each row. In general, each successive row has a delay which is \(J\) symbols duration higher than the previous row as shown in Fig.3. The code word symbol from the encoder is fed into the array of shift registers, one code symbol to each row. With each new code word symbol the commutator switches to a new register and the new code symbol is shifted out to the channel. The \(i\)-th (\(1 \leq i \leq N-1\)) shift register has a length of \((i-1)J\) stages where \(J = M/N\) and the last row has \(M-1\) numbers of delay elements. The convolutional deinterleaver performs the inverse operation of the interleaver and differs in structure of the arrangement of delay elements. Zeroth row of interleaver becomes the \(N-1\) row in the deinterleaver. 1st row of the former becomes \(N-2\) row of later and so on.

Fig.3: Convolution Interleaver.

In order to verify the Verilog HDL models for the interleaver and deinterleaver the authors have developed another top level Verilog HDL model, combining interleaver and deinterleaver. The scrambled code words from the output of the interleaver is applied as input to the
deinterleaver block along with clock as synchronization signal. It is observed in Fig 4 that the scrambled code word is converted into its original form at the output of the deinterleaver block.

Fig 4: Block diagram of 8 bit Convolutional Interleaver.

V. SOFTWARE ANALYSIS OF TURBO DECODER

Simulink model of a Turbo Encoder and Turbo Decoder are shown below.

A) Turbo Encoder:

It consists of two convolutional encoders. The outputs of the turbo encoder are the information sequence, together with the corresponding parity sequence produced by first encoder and the parity sequence produced by the second encoder block, the input to second encoder is through interleaver, which scrambles the data bit sequence. Simulation model of Turbo encoder - decoder is shown in Fig.5.

B) Turbo Decoder:

The proposed Turbo decoder shown above in Fig 6 uses iterative decoding. The turbo code decoder is based on a modified Viterbi algorithm that incorporates reliability values to improve decoding performance. The turbo decoder consists of M-elementary decoders one for each encoder in turbo encoding part. Each elementary decoder uses the Soft Decision Viterbi Decoding to produce a soft decision for each received bit. After an iteration of the decoding process, every elementary decoder shares its soft decision output with the other M-1 elementary decoders.

C) Channel:

The AWGN Channel block adds white Gaussian noise to a real or complex input signal. Each of the major blocks mentioned above have individual sub blocks which are configured to meet the Specifications (After scaling, keeping in mind the mathematical constraints of modelling a real time system).

VI. WORK FLOW

Simulink model for turbo encoder and Turbo decoder is developed on MATLAB platform. The Verilog HDL coding is made on Modelsim and verified. The HDL codes are synthesized by Synopsys Design Compiler. Fig 7 shows the work flow for implementation.

Fig 7: Work flow for implementation
VII. CONCLUSIONS

In this paper, presented a brief survey about turbo codes, designed a turbo encoder and turbo decoder on simulink and coded in Verilog HDL. The codes and coding techniques are carried by Viterbi decoder. For turbo codes, turbo decoders and their decoding algorithms like log-MAP/SOVA are used. Based on the algebraic constructions, the interleaver offers capability which enables Turbo decoding by using MAP decoders working concurrently. We proposed low complexity recursive architecture for generating the convolutional interleaver addresses on the fly. The convolution interleavers are designed to operate at full speed with the MAP decoders. The proposed architecture has been scaled and can be tailored for different throughput requirements.

ACKNOWLEDGMENT

The authors would like to thank Manjunatha K N and Prasanna Kumar Kfor their research papers as we referred their research papers for our work. They would also like to thank Prof. Salim A. Chavan for their valuable guidance regarding this paper work. The authors would also extend their sincere thanks to all the Teaching staff from Electronics & Telecommunication department of Dr. Bhausaheb Nandurkar College of Engineering & Technology, for many interesting discussions on this topic.

REFERENCES

[5] Sandro Belfanti, Christoph Roth, Michael Gautschi, Christian Benkeser, and Quting Huang.”A 1Gbps LTE-Advanced Turbo-Decoder ASIC in 65nm CMOS.”Integrated Systems Laboratory, ETH Zurich, 8092 Zurich, Switzerland.
Cloud Computing; Approach towards Infrastructure, Service Models

1Prof Anuja k. Pande, 2Rohit S. Bhore, 3Sejal Bharkhada, 4Ashwini Malik, Prof Trupti Lokhande.
6Computer Engineering, DBNCOET, Amravati University
Yavatmal, India.

Abstract— Cloud Computing Is one of the computing model, not a technology. Cloud computing is a new paradigm in which computing resources such as processing, memory, and storage are not physically present at the user’s location. Instead, a service provider owns and manages these resources, and users access them via the Internet. Cloud computing promises a more cost effective enabling technology to outsource storage and computations. Cloud computing (CC) has been widely recognized as the next generation’s computing infrastructure. CC offers some advantages by allowing users to use infrastructure (e.g., servers, networks, and storages), platforms (e.g., middleware services and operating systems), and software’s (e.g., application programs) provided by cloud providers (e.g., Google, Amazon,) at low cost. In addition, CC enables users to elastically utilize resources in an on-demand fashion. As a result, mobile applications can be rapidly provisioned and released with the minimal management efforts or service provider’s interactions. The term “Cloud” is analogical to “Internet”. The cloud computing is Internet based computing where virtual shared server provides software, infrastructure, platform, devices and other resources. All information that a digitized system has to offer is providing as a services in the cloud computing model. User can access these services available on the Internet Cloud without having any previous know-how managing the resources involved. Thus, user can concentrate more on their core business processes rather than spend time and gaining knowledge on resources that needed to manage on cloud.

Keywords— Cloud Computing, Infrastructure, Service Model, Security.

X. INTRODUCTION

The term “Cloud” is analogical to “Internet”. The cloud computing is Internet based computing where virtual shared server provides software, infrastructure, platform, devices and other resources. All information that a digitized system has to offer is providing as a services in the cloud computing model. User can access these services available on the Internet Cloud without having any previous know-how managing the resources involved. Thus, user can concentrate more on their core business processes rather than spend time and gaining knowledge on resources that needed to manage on cloud. Cloud computing is a colloquial expression used to describe a variety of different types of computing concepts that involve a large number of computers that are connected through a real-time communication network (typically the Internet).

Cloud computing is a type of computing that relies on sharing computing resources rather than having local servers or personal devices to handle applications. In cloud computing, the word cloud (also phrased as "the cloud") is used as a metaphor for "the Internet," so the phrase cloud computing means "a type of Internet-based computing," where different services -- such as servers, storage and applications -- are delivered to an organization's computers and devices through the Internet.

Fig 1.1 Cloud computing logical diagram

XI. CLOUD COMPUTING FEATURES

Cloud Computing Is one of the computing model, not a technology. Cloud computing is a new paradigm in which computing resources such as processing, memory, and storage are not physically present at the user’s location.

2.1 Why Cloud computing?

Cloud computing is an industry transformation. Cloud computing enables businesses, of all sizes to deliver IT as a service, offering new possibilities to focus more on business success and less on operational costs and maintenance.

There are many advantages a user can leverage from cloud computing. They are listed as follows,

- Cloud computing user avoids capital expenditure on building up an infrastructure to support their
application. Instead, they pay the provider only the amount they consume.

- The user need not invest on the maintenance of the Infrastructure of the application. The provider maintains the infrastructure for the user.
- The user can access the multiple data servers from any location at a go.
- Enhancement of the application is easy, as the user need not worry about the infrastructure enhancement.
- Cloud computing is an eco-friendly incentive which will replace the hardware components with services.

2.2 Cloud computing Features

Cloud Computing brings features that distinguish it from classical resource and service provisioning environments, they are as follows,

2.2.1 Highly Scalable—Cloud computing provides resources and services for users on demand. The resources are scalable over several data centers.

2.2.2 Less capital expenditure—Cloud computing does not require upfront investment. No capital expenditure is required. Users may pay and use or pay for services and capacity as they need them.

2.2.3 Higher resource Utilization - Cloud computing can guarantee QoS for users in terms of hardware or CPU performance, bandwidth, and memory capacity.

2.2.4 Disaster recovery and Back up.

2.2.5 Device and Location Independence.

XII. CLOUD COMPUTING SERVICE MODELS

![Cloud Computing Service Models](image)

**Software as a Service (SaaS)**

It is a model of software deployment whereby the provider licenses an application to the customers for use as a service on demand. The capability provided to the End users is to use the provider’s applications running on a cloud infrastructure. The applications are accessible from various client devices through a thin client interface such as a web browser (e.g., web enabled e-mail). The end users does not manage or control the underlying cloud infrastructure including network, servers, operating systems, storage, or even individual application capabilities, with the possible exception of limited user specific application configuration settings. Today SaaS is offered by companies such as Google, Sales force, Microsoft, Zoho, etc.

**Platform as a Service (PaaS)**

It is the delivery of computing platform and solution stack as a service. The capability provided to the end users is to deploy onto the cloud infrastructure user created or acquired applications created using programming languages and tools supported by the provider. The end user does not manage or control the underlying cloud infrastructure including network, servers, operating systems, or storage. PaaS provides all the facilities required to support the complete life cycle of building and delivering web applications entirely on the web. As Platform-as-a-Service (PaaS) is available as a service, the developer and ISV’s get full control of the application development and deployment. PaaS enables developers and ISV’s to create custom web applications and deliver it quickly, as many of the hassles like setting up hosting, servers, databases, user interaction process and frameworks are pre-packaged. PaaS a concept is known as Platform as a Service and also as Cloud Computing Platform. PaaS applications are referred as On-Demand, Web-based or as Software as a Service (SaaS) Applications.

**Infrastructure as a Service (IaaS)**

It is the delivery of computer infrastructure (typically a platform virtualization environment) as a service. The capability provided to the end users is to provision processing, storage, networks, and other fundamental computing resources where the end user is able to deploy and run arbitrary software, which can include operating systems and applications. The user does not manage or control the underlying cloud infrastructure but it has control over operating systems, storage, deployed applications, and possibly limited control of select
networking components. Some of the common examples are Amazon, Go Grid, 3tera, etc.

With “Infrastructure as a Service” (IaaS), customers get on-demand computing and storage to host, scale, and manage applications and services. IaaS delivers computer infrastructure—a typically a platform virtualization environment—as a service. Rather than purchasing servers, software, data-centre space and network equipment, customers buy those resources as fully outsourced services. Suppliers typically bill such services based on a utility computing basis and amount of resources consumed—therefore the cost will typically reflect the level of activity.

**Public Clouds**

Public clouds are run by third parties, and applications from different customers are likely to be mixed together on the cloud’s servers, storage systems, and networks. Public clouds are most often hosted away from customer premises, and they provide a way to reduce customer risk and cost by providing a flexible, even temporary extension to enterprise infrastructure. If a public cloud in implemented with performance, security, and data locality in mind, the existence of other applications running in the cloud should be transparent to both cloud architects and end users. Indeed, one of the benefits of public clouds is that they can be much larger than a company’s private cloud might be, offering the ability to scale up and down on demand, and shifting infrastructure risks from the enterprise to the cloud provider, if even just temporarily. Portions of a public cloud can be carved out for the exclusive use of a single client, creating a virtual private datacenter. Rather than being limited to deploying virtual machine images in a public cloud, a virtual private datacenter gives customers greater visibility into its infrastructure. Now customers can manipulate not just virtual machine images, but also servers, storage systems, network devices, and network topology. Creating a virtual private datacenter with all components located in the same facility helps to lessen the issue of data locality because bandwidth is abundant and typically free when connecting resources within the same facility.

**Private Clouds**

Private clouds are built for the exclusive use of one client, providing the utmost control over data, security, and quality of service. The company owns the infrastructure and has control over how applications are deployed on it. Private clouds may be deployed in an enterprise datacenter, and they also may be deployed at a co-location facility. Private clouds can be built and managed by a company’s own IT organization or by a cloud provider. In this “hosted private” model, a company such as Sun can install, configure, and operate the infrastructure to support a private cloud within a company’s enterprise datacenter. This model gives companies a high level of control over the use of cloud resources while bringing in the expertise needed to establish and operate the environment.

**Hybrid Clouds**

Hybrid clouds combine both public and private cloud models. They can help to provide on-demand, externally provisioned scale. The ability to augment a private cloud with the resources of a public cloud can be used to maintain service levels in the face of rapid workload fluctuations. This is most often seen with the use of storage clouds to support Web 2.0 applications. A hybrid cloud can also be used to handle planned workload spikes.
Sometimes called “surge computing,” a public cloud can be used to perform periodic tasks that can be deployed easily on a public cloud. Hybrid clouds introduce the complexity of determining how to distribute applications across both a public and private cloud. Among the issues that need to be considered is the relationship between data and processing resources. If the data is small, or the application is stateless, a hybrid cloud can be much more successful than if large amounts of data must be transferred into a public cloud for a small amount of processing.

XIV. APPLICATIONS OF CLOUD COMPUTING

Social Networking

Perhaps the most famous use of cloud computing, which does not strike people as “cloud computing” at first glance is social networking Websites, including Facebook, LinkedIn, MySpace, Twitter, and many, many others. The main idea of social networking is to find people you already know or people you would like to know and share your information with them. Of course, when you share your information with these people, you’re also sharing it with the people who run the service. While the primary purpose of social networking previously was connecting people, businesses can use social networking too. By creating a Facebook fan page, a business can connect with its customers, and at the same time, those customers will be promoting your business. Also, viral marketing tactics can be used in combination with social networks. There are public relations experts who specialize in social media marketing.

Backup Services

Even if you do use services to keep all your documents and photos, chances are you still have data on your personal computer. One of the biggest problems with personal computing has been the tendency to lose that data if your computer is stolen, destroyed, or the storage device damaged. This is where backup comes in. Sometimes, even backing up to media you have isn’t good enough -- you need to store the data off-site for more complete protection. Services like JungleDisk, Carbonite, and Mozy allow you to automatically back up all your data to servers spread around the country or world for a surprisingly low price. Of course, your data is then susceptible to security breaches. Similarly, services like Syncplicity and Dropbox (both offer free versions) make it easy to keep local copies of files on multiple computers synchronized while keeping a copy in the “cloud.” Some of these services will even keep previous versions of files or deleted files in case you happen to delete or mess up an important file.

Banking and Financial Service

Consumers store personal financial information to cloud computing service providers. In addition, consumers store tax records using free or low cost online backup services.

Health Care

In an effort to improve the nation’s health IT infrastructure, the Department of Health and Human Services’ (HHS) Office of the National Coordinator for Health Information Technology (ONC) recently selected a cloud computing platform to manage the selection and implementation of Electronic Health Record (EHR) systems across the country. Non-health care organizations as Google and Microsoft provide a means by which consumers can create an online personal health record (“PHR”). Google Health and Microsoft Health Vault allow the public to create, store, and access online personal health records on the search engine's website.

XV. ADVANTAGES OF CLOUD COMPUTING

There are many advantages of storing the data on cloud. Cloud computing also provides much application that is advantages of cloud. The advantages of cloud are listed as below:

Privacy

The main advantage of cloud, it maintains the privacy and security of user data. Cloud computing poses privacy concerns because the service provider can access the data that is on the cloud at any time. It could accidentally or deliberately alter or even delete information. Privacy advocates have criticized the cloud model for giving hosting companies' greater ease to control—and thus, to monitor at will—communication between host company and end user, and access user data (with or without permission). Instances such as the secret NSA program, working with AT&T, and Verizon, which recorded over 10 million telephone calls between American citizens, causes uncertainty among privacy advocates, and the greater powers it gives to telecommunication companies to monitor user activity. A cloud service provider (CSP) can complicate data privacy because of the extent of virtualization (virtual machines) and cloud storage used to implement cloud service. CSP operations, customer or tenant data may not remain on the same system, or in the same data center or even within the same provider's cloud; this can lead to legal concerns over jurisdiction. While there have been efforts (such as US-EU Safe Harbor) to "harmonies" the legal environment, providers such as Amazon still cater to major markets (typically the United States and the European Union) by deploying local infrastructure and allowing customers to select "availability zones."
Security
As cloud computing is achieving increased popularity, concerns are being voiced about the security issues introduced through adoption of this new model. The effectiveness and efficiency of traditional protection mechanisms are being reconsidered as the characteristics of this innovative deployment model can differ widely from those of traditional architectures. An alternative perspective on the topic of cloud security is that this is but another, although quite broad, case of “applied security” and that similar security principles that apply in shared multi-user mainframe security models apply with cloud security. The relative security of cloud computing services is a contentious issue that may be delaying its adoption. Physical control of the Private Cloud equipment is more secure than having the equipment off site and under someone else’s control. Physical control and the ability to visually inspect data links and access ports is required in order to ensure data links are not compromised. Issues barring the adoption of cloud computing are due in large part to the private and public sectors’ unease surrounding the external management of security-based services. It is the very nature of cloud computing-based services, private or public, that promote external management of provided services. This delivers great incentive to cloud computing service providers to prioritize building and maintaining strong management of secure services. Security issues have been categorized into sensitive data access, data segregation, privacy, bug exploitation, recovery, accountability, malicious insiders, management console security, account control, and multi-tenancy issues. Solutions to various cloud security issues vary, from cryptography, particularly public key infrastructure (PKI), to use of multiple cloud providers, standardization of APIs, and improving virtual machine support and legal support.

Legal
As with other changes in the landscape of computing, certain legal issues arise with cloud computing, including trademark infringement, security concerns and sharing of proprietary data resources. If a cloud company is the possessore of the data, thepossessor has certain legal rights. If the cloud company is the “custodian” of the data, then a different set of rights would apply. The next problem in the legalities of cloud computing is the problem of legal ownership of the data. Many Terms of Service agreements are silent on the question of ownership. These legal issues are not confined to the time period in which the cloud based application is actively being used. There must also be consideration for what happens when the provider-customer relationship ends. In most cases, this event will be addressed before an application is deployed to the cloud. However, in the case of provider insolvencies or bankruptcy the state of the data may become blurred.

Improving data storage capacity and processing power
Storage capacity is also a constraint for computing devices. Cloud computing is developed to enable users to store/access the large data on the cloud through wireless networks. First example is the Amazon Simple Storage Service (Amazon S3) which supports file storage service. Another example is Image Exchange which utilizes the large storage space in clouds for mobile users. This mobile photo sharing service enables mobile users to upload images to the clouds immediately after capturing. Users may access all images from any devices. With cloud, the users can save considerable amount of energy and storage space on their mobile devices since all images are sent and processed on the clouds. Facebook is the most successful social network application today, and it is also a typical example of using cloud in sharing images. Mobile applications also are not constrained by storage capacity on the devices because their data now is stored on the cloud.

XVI. LIMITATIONS OF CLOUD COMPUTING
Although highly promising, cloud computing for multimedia applications is still in its infancy. There are many unsolved problems need to be investigated to fully harvest its potential. Cloud Computing relies on wireless networks (e.g., 3G and Wi-Fi) for data and control between the cloud and other computing devices. Compared with fixed and wired networks, wireless networks have limited bandwidth, probably longer latency, and intermittent connectivity. Moreover, under the presence of more mobile devices, the bandwidth available to each device will be further reduced, and network latency can go up and response time for wireless network users can be larger. Following are the some of the limitations of Cloud Computing.

Availability
Various network failures and traffic congestion in the network may hinder the mobile users to connect to the cloud and access required services. If there are too many concurrent requests, even if the powerful cloud servers are capable of serving these requests, the wireless links will surely fall short of capacity.

Bandwidth Costs
Cloud computing, companies can save money on hardware and software; however they could incur higher network bandwidth charges. Bandwidth cost may be low for smaller Internet-based applications, which are not data intensive, but could significantly grow for data-intensive applications.

Requires a constant Internet connection
- Cloud computing is impossible if you cannot connect to the Internet.
- Since you use the Internet to connect to both your applications and documents, if you do not have an
Internet connection you cannot access anything, even your own documents.

- A dead Internet connection means no work and in areas where Internet connections are few or inherently unreliable, this could be a deal-breaker.

XVII. CONCLUSION

Cloud Computing is one of the computing model, not a technology. Cloud computing is a new paradigm in which computing resources such as processing, memory, and storage are not physically present at the user's location. Instead, a service provider owns and manages these resources, and users access them via the Internet. Cloud computing promises a more cost effective enabling technology to outsource storage and computations.

The applications supported by cloud computing including social network, backup services, banking and financial, healthcare have been discussed which clearly show the applicability of the mobile cloud computing to a wide range of services. Although devices geared toward cloud computing will undoubtedly change technology trends as well as our daily lives, some practical problems remain to be resolved to structure a full cloud system.

ACKNOWLEDGMENT

We acknowledge our senior faculty who have provided us their views in the selection of topic.

REFERENCES

[22] Andrieux, A., Czajkowski, K., Dan, A., Keahey, K., Ludwig, H., Pruyne, J.,
[23] Rofrano, J., Taecke, S., Xu, M.: Web services agreement specification (WS-
[27] Department, University of California, Berkeley (Feb 2009)
Rapid Information Collection from a Wireless Sensor Network Organised as Tree

Nandkishor N. Bende¹, Mayur V. Guhe², A. W. Motikar³

¹B.E., Electronics & Telecommunication Engg,
²B.E., Electronics & Telecommunication Engg,
³Assistant prof. DBNCOET, Yavatmal

Sant Gadge Baba Amravati University, Amravati DBNCOET Yavatmal, Maharashtra (INDIA)

Abstract- Fast and energy efficient data collection in an energy constraint ad-hoc sensor network is always a challenging issue. The network topology and interferences causes significant effects on data collection and hence on sensors’ energy usage. Various approaches using single channel, multichannel and convergecasting had already been proposed. Here in this paper we have shown data collection performance using multi-frequency in channel assignment, and effect of network topology, for moderate size networks of about 50-100 nodes. For the study we have used some realistic simulation models under many-to-one communication paradigm called convergecast, a single frequency channel and TDMA technique to have minimum time slots for convergecasting.

Key Terms: - Convergecast; Multi-channel; Topology; Energy; TDMA

I. INTRODUCTION

A tree base sensor network is a collection of sensors nodes, such as sink is the root of tree and leaves are the nodes. Data in such topology flows from sensor nodes (leaves) to the sink (root) of the tree. Collection of data from a set of sensors to an intermediate parent (sink) in a tree is known as converge-casting. The ‘delivery-time’ and ‘data-rate’ are application specific. As an example, in oil and gas refineries the sensor devices and controllers need to collect data from all the sensors within a specific deadline [1] for any kind of leakage or failures. Whereas applications like weather forcasting, under-water observations needs continuous and fast data delivery for analysis, for longer periods. Here in this paper our emphasis is on such applications focusing on fast data streaming from sensor to sink node. The two common approaches for data collection [3] are - 'aggregated-data convergecast' where packets are aggregated at each hop, and 'raw-data convergecast' where each data packet travel towards sink node individually. First approach is most suitable where data is highly co-related and objective is to collect maximum sensor reading and second approach is used where the reading of each sensor is equally important. Further, interference and network topology are the two prime limiting factors in wireless sensor networks. Time Division Multiple Access (TDMA) [2] protocol is well suited to periodic data traffic to have contention free medium and to avoid collisions. The use of multiple frequency channels can allow more concurrent transmissions. Here we have shown that if multiple frequencies are employed along with TDMA, the data collection rate is affected by tree topology and not by interferences. Thus, in this paper we identify the effect of network topology on the schedule length, and analyzed the performance of convergecast by using multiple frequencies as compared to those trees using a single frequency.

The rest of the paper is organized as follows: in Section II, we discuss related works. In Section III, we describe system modeling and some discussions. In Section IV, we have shown multichannel scheduling for interference elimination. In Section V, we focus on impact of network topology on data forwarding. Section VI gives the evaluation work based on previous discussions. Finally Section VII concludes the paper.

II. RELATED WORK

Gandham et al. [2] proposed a distributed time slot assignment scheme, for a single channel in TDMA schedule length. Fast data collection with minimum schedule length for aggregated convergecast is discussed by Chen et al. in [3]. Annamalai et al. [4] uses the concept of orthogonal codes to remove interferences, where each node has been assigned time slots from bottom to the top of the tree such that a parent has to wait till it receives all the data packet from its children. Pan and Tseng [5] described a beacon period, assigned to every sensor node in Zigbee network, scheme to reduce latency. A node can receive data only in the allotted beacon period. Song et al.[6] described a time-optimal energy efficient packet scheduling algorithm for raw-data convergecast with periodic traffic. They assumed a simple interference model in which every node has a circular transmission range and interferences from concurrent multiple senders is neglected. Song et al. [6] further extended their work and proposed a TDMA- based MAC protocol for high-
data-rate [7]. Choi et al. [8] shows that for a single channel the minimum schedule length for raw-data convergecast is NP complete on general graphs. Lai et al. [9] uses a greedy graph coloring approach to find the shortest path to the senders for throughput optimization. They also focused on impact of routing trees on schedule length and devised a new approach called disjoint strips to transmit data over different shortest paths. The use of multiple frequencies is widely described in [10] and [11].

III. SYSTEM MODELLING AND DISCUSSION

Let \( G = (V, E) \) is a multi-hop WSN graph, where \( V \) is the set of sensor nodes, and \( E = (I, j) : (I, j) \in V \) is the set of wireless links. Let \( s \) is the sink node such that \( S \subseteq V \). The distance between two nodes \( I \) and \( j \) is denoted by \( D(I, j) \). All the nodes other than \( S \) generate and transmit data packets through a network path to sink \( s \). Let, \( T = (V, i \in E) \) is a spanning tree on \( G \) where \( E \) and \( E \) represents the tree edges. It is assumed that each node has half-T duplex transceiver; therefore it cannot simultaneously send and receive data. We have used equal sized time frame TDMA protocol and two types of interference models for analysis namely; SINR based physical model and graph based model. The interference range of a node is equal to its transmission range which means two links cannot be scheduled at the same time if receiver of one link is within the transmitter range of the other link. In SINR model the successful reception of the packet from \( i \) to \( j \) depends on cumulative interference caused by all concurrent transmitting nodes and the ratio between the received signal strength at \( j \). The size of each data packet is assumed to be same. For fast data routing we aim to schedule the edges \( E \) of \( T \) using a minimum \( T \) number of time slots with two constraints:

**Adjacency constraint:**

It states that two edges in \( E \) cannot be scheduled in same time slot if they are adjacent \( T \) to each other. This is because of half duplex transceiver available on nodes.

**Interfering constraint:**

The interfering constraint depends on the choice of the interference model. For a periodic data collection in aggregated convergecast each edge in \( E \) is scheduled in a pipeline manner. The sink receives packets from the pipeline one after another. On the other hand, for raw data convergecast the edge in \( E \) is scheduled multiple times hence no pipeline is there.

A. Raw-Data Convergecast

In it data of each sensor is equally important, therefore aggregation is not desirable. Each packet is individually scheduled to reach sink node. The problem of minimizing the scheduling length for raw-data convergecast is proved to be NP-complete. Fig. 1, shows a basic tree structure where \( \{s, 1, 4\}, \{s, 2, 5\}, \{s, 3, 6\} \) are branches of tree and \( \{1, 4\}, \{2, 5\}, \{3, 6\} \) are sub-trees. We can deduce a local time slot allotment algorithm for each node with an objective to schedule parallel transmissions and allow sink to collect data packets continuously. We assume that sink knows the number of available nodes in each top sub-trees. Each node maintains buffer and state of full or empty if it has data packet available or not. The algorithm for raw data convergecast slot allotment is shown in Algorithm 1.

![Fig:3.1 Tree Topology](http://www.ijettjournal.org)

1.Algorithm 1: Local Time Slot Scheduling

1. Initialize node[buffer]=FULL
2. Pick a node (N)
3. If (N = Sink) then
   Among available sub-tree, select one with largest number of remaining packets (i).
4. Plan a link (root(i), S)
5. Else IF (N != Sink and node[buffer] = EMPTY) then
6. Select a child (C) at random whose buffer is full
7. Plan a link (C, node)
8. C[buffer]= EMPTY
9. End If
10. End If

In Algorithm 1 lines 3-4 gives scheduling rules between sink and root of sub trees. A top subtree is eligible to be elected for transmission if it has at least one packet for transmission. If none of the top- sub trees are eligible, the sink does not receive any packet during that time slot. Inside each top-sub tree, nodes are scheduled according to the rules in lines 5-8. We define a sub tree to be active if there are still packets left in the sub tree to be relayed. If a node’s buffer is empty and the sub tree rooted at this node is active, we schedule one of its children at random whose buffer is not empty.
B. Aggregated Data Convergecast

For continuous monitoring applications data aggregation technique is most suitable. It helps in removing data redundancy and minimizes count of transmission, thus saves energy [12]. Aggregation can be achieved by different techniques such as by data compression, suppressing duplicate messages or by packet merging technology etc. The size of aggregated data transmitted is same and does not depend on to the size of data on individual sensor. The examples of such aggregation functions are MIN, MAX, MEDIAN, AVERAGE, etc.

Algorithm 2: Aggregation Tree Algorithm
1. Start
2. Let T = (V, E_T )
3. while E_T is not EMPTY do Select edge (e) from E using Breadth First Search Manner
4. Allocate minimum time slot t to the selected edge e
5. Move to next edge in E_T
6. End

IV. MULTICHANNELSCHEDULING FOR INTERFERENCEELIMINATION

Multichannel communication allows concurrent transmissions by using different frequencies [13], hence eliminate interference. Although typical WSN radios operate on a limited bandwidth, their operating frequencies can be adjusted. It enables multiple concurrent transmissions and more data delivery. By assuming fixed bandwidth channels, we explain two channel assignment methods and study their pros and cons for both type of convergecast. These methods are link level (JFTSS) and cluster level (TMCP). Joint Frequency Time Slot Scheduling (JFTSS) enables a greedy joint solution for maximal time schedule. A maximal schedule is that which meets the adjacency and interfering constraints, and no further links can be scheduled for concurrent transmissions on any time slot. A comparative study of single channel and multichannel system is discussed in [14]. JFTSS scheduling in a network starts from the link having highest number of packets for transmission. If the link loads are equal, the most constrained link is opted first. Initially algorithm has an empty schedule and links are sorted as per loads. The links having adjacency constraint with scheduled link are excluded from the list of link to be scheduled in a given time slot. Only the link having non-interfering constraint with scheduled link can be scheduled in the same slot and those having interfering constraint can be scheduled on different channels. If no more links are possible to be scheduled for a given slot, the scheduler continues with scheduling in the next slot.

Fig. 3.2 Aggregated Convergecast Pipeline

In Fig. 2(a) and 2(b) aggregated convergecast pipeline is shown for five nodes. The solid lines represent tree edges, and the dotted lines represent interfering links. The numbers beside the links represent the time slots at which the links are scheduled to transmit. The numbers inside the circles denote node ids. The table shows the list of senders and receiver in each time slot. Here at the end of Frame 1 the sink has no data packets from node 5, as the schedule is repeated, it receives aggregated packets from 2 and 5 in next frame slot. The entries {1, 4} and {2, 5} in the table shows single packets comprising aggregated data from nodes 1 and 4, and from nodes 2 and 5 respectively. Therefore after Frame 2 a pipeline is generated and sink receives aggregated packets from all the nodes. Now a time slot allotment algorithm can be generated and is shown in Algorithm 2.

Fig:4.1 JFTSS scheduling

Fig. 4.1 shows the same tree in Fig.3.1(a) which is scheduled according to JFTSS to collect aggregated data.
JFTSS starts with link (2, s) on frequency 1 (F1) and then schedules link (4, 1) on the first slot on frequency 2 (F2). Then, links (5, 2) on frequency 1 (F1) and (1, s) on frequency 2 (F2) are scheduled on the second slot and (3, s) on frequency 2 (F2) are scheduled on the last slot. An advantage of JFTSS is that it is easy to incorporate the physical interference model; however, it is hard to have a distributed solution since the interference relationship between all the links must be known.

Tree Multi-Channel Protocol (TMCP) [10] is a greedy tree-based multichannel protocol. It divides a complete tree into number of sub-trees and reduces intra tree interference by using different channels to the nodes

Fig:4.2 TMCPscheduling

Fig. 4.2 shows the same tree given in Fig.3.2a, scheduled according to TMCP. Here, the nodes on the leftmost branch are assigned frequency F1, second branch is assigned frequency F2, and the last branch is assigned frequency F3 and after the channel assignments, time slots are assigned to the nodes with algorithm 2. The advantage of TMCP is that it is designed to support convergecast traffic and does not require channel switching. Since all the nodes communicate on same channel, the contention inside branches is not resolved.

V. IMPACT OF NETWORK TOPOLOGY

Besides multiple channels, the network topology and the degree of connectivity makes impact on scheduling performance. As described in [8], network trees that have more parallel transmissions do not necessarily result in small schedule lengths. As an example in star topology network with N nodes the schedule length is N, whereas it is (2N-1) for a bus topology once interference is eliminated. In this section we construct a spanning tree with constraint n < (N+1)/2, where n are number of branches and N is the number of nodes. A balanced k tree satisfying this constraint is a variant of a capacitated minimal spanning tree (CMST) [15]. The CMST algorithm can determine a minimum-hop spanning tree in a vertex weighted graph, such that the weight of every subtree linked to the root does not exceed a prescribed capacity. Here we have taken weight of each link as 1, and prescribed capacity is (N+1)/2. Here, we propose a method, described in Algorithm 3, based on greedy scheme presented by Dai and Han [16] to solves a variant of the CMST problem. By using it, searches for routing trees with an equal number of nodes on each branch. It is assumed that every node know their minimum-hop counts to sink node.

Algorithm 3: CMST Tree Creation

1. Given G(V, E) with sink node S
2. Let P is roots of top subtrees and T={s} U P; k=2; RS(i)=unconnected neighbors of I; S(i)=0;
3. while k != Maximum_hop_count do
   N_h=all unconnected nodes at hop distance h;
   Connect node N' having single parent: T=T U N_h';
4. Update N_h=N_h \ N_h';
5. Sort N_h;
6. for all i in N_h do
   For all j in P to which i can connect do Link (i, j);
   End for T=T U {i} U Link(i, j); Update RS(i); End for
7. k=k+1;
8. End while

The rules associated to the algorithm are:

Rule 1.
Nodes having single parents are connected first.

Rule 2.
Node with multiple parents, a Reservoir Set (RS) is created and selects one from it.

Rule 3.
After selecting a node from RS a search set S is constructed to decide which particular branch the node should be added to. S therefore consists of nodes that are not yet connected but are neighbors of a node with high hop-count.

VI. PERFORMANCE EVALUATION

In this section, we evaluate performance of multiple channels and network tree topology on scheduling for both aggregated and raw-data convergecast. We deploy sensor nodes randomly in a region with dimensions varying between 30x30 m and 400x400 m to have different network density. The number of nodes is fixed to 100 and for different parameters; we
average each point over 1500 runs. An exponential path-loss model for signal propagation along with path-loss exponent varying between 3 and 4 is used. We have simulated the behavior of CC2420 radios used on TmoteSky motes and are able to operate on 16 different frequencies. The transmission power can be adjusted between -24 and 0 dBm over eight different levels and the SINR threshold is set to = -3dB. Firstly, the schedule length of single-channel TDMA is determined, secondly its improvement using multiple channels and routing trees is evaluated. All the nodes transmit at maximum power and uses minimum hop tree. In TMCP time slots are assigned according to Algorithm 1 for raw data convergecast and Algorithm 2 for aggregated convergecast. The path loss exponent is 3.5. The results are shown in Fig. 5(a) and 5(b). It is evident from Fig. 5(a) that with just two frequencies interference limitations are eliminated and the performance gains are limited by the connectivity structure. With multichannel communication a 40 percent reduction in schedule length is observed as compared to transmitting on a single channel with maximum power. Further, JFTSS can optimally schedule the network using 16 channels as shown in graph of fig.5. In dense deployments, TMCP performs better due to construction of different routing trees i.e., when L = 20, JFTSS construct a star topology, whereas TMCP constructs a 2-branch tree with two channels and a 16-branch tree with 16 channels.

VII. CONCLUSION

In this paper, we have discussed fast convergecast methods in wireless sensor network, where nodes communicate using TDMA protocol so as to minimize the scheduling length. We have focused on fundamental shortcoming because of interference and half duplex transceivers available on the nodes. We observed that multiple channel method is helpful in reducing schedule length. We also determined that link-based (JFTSS) channel assignment schemes are more energy efficient in removing interference, if compared to TMCP scheduling schemes. Through extensive simulations, we demonstrated up to certain extent reduction in schedule length for aggregated data convergecasting and approximately 50 percent reduction for raw-data convergecast. As a future work we will explore aspects related to variable amount of data and evaluate the various schemes considered.

REFERENCES


Patient Monitoring System using Wireless Technology

Priyanka More¹, Pragati Daroi², Amol Alkari³

¹Final Year B.E., Dept of EnTC, DBNCOET, Yavatmal (MS), India.
²Final Year B.E., Dept of EnTC, DBNCOET, Yavatmal (MS), India.
³Asstt. Prof., Dept of EnTC, DBNCOET, Yavatmal (MS), India.

Abstract— In today's fast phase of life where human population is increasing and their health problems are also increasing with them. Every day we see so many road accident, people are suffering from so many diseases. This increases the crowd in the hospital. Development of a system for monitoring of multiple subject’s physiological parameters in a hospital on a single monitor in doctors cabin and subjective workload regardless of location has been presented. This allows an occupational health and also provides the necessary platform to monitor, analyze & control the system. For this purpose modern acquisition systems are needed. A sensor electronics module permits the acquisition of different physiological parameters of different subjects on a single screen and their online transmission to the other doctors in case of need. The sensor electronics module constitutes a wireless personal area network so that real time monitoring is possible & patient can be helped on time. This electronic input of multiple patients is better than manual documentation on papers.

Keywords— GPRS, sensor network, wireless network, data acquisition.

I. INTRODUCTION

In this paper the development of a system for monitoring of multiple subjects physiological parameters in a hospital on a single monitor in doctors cabin is presented. In view of the ever-growing age median among travelers, a health monitoring application is becoming more of a necessity in large capacity aircraft environments, providing safety to passengers with actual or chronic risks, and reducing risk and cost for Long-range aircraft operations. Considering the technological advancements in embedded sensor devices a portable medical monitoring enclosure has been developed to provide with the flexibility of low Cost and high accuracy measurement equipment in avionic environments. The patient physical states data acquisition and communication system monitors the main physical parameters & movement status continuously. The information from data acquisition system is sent to other hospital centers by wireless communication module. The monitoring centers receive the information from each patient and transmit it through GPRS technology. The data from patient can be displayed as numeric or graph on monitor & then the doctor can diagnose the patient according to recorded continuous data. A sensor electronics module permits the acquisition different physiological parameters so that real time monitoring is possible & is helpful in worst condition. If any patient is serious the red mark blinks on the monitor, which takes attention of doctor on that.

II. LITERATURE REVIEW

In 1625, Santorio, who lived in Venice at the time, published his methods for measuring body temperature with the spirit thermometer and for timing the pulse (heart) rate with a pendulum. The principles for both devices had been established by Galileo.

In 1903, Willem Einthoven devised the string galvanometer for measuring the ECG, for which he was awarded the 1924 Nobel Prize in physiology. The ECG has become an important adjunct to the clinician's inventory of tests for both acutely and chronically ill patients. Continuous measurement of physiological variables has become a routine part of the monitoring of critically ill patients.

At the same time advances in monitoring were made and major changes in the therapy of life-threatening disorders were also occurring. Prompt quantitative evaluation of measured physiological and biochemical variables became essential in the decision making process as physicians applied new therapeutic interventions. For example, it is now possible and in many cases essential to use ventilators when a patient cannot breathe independently, cardiopulmonary bypass equipment when a patient undergoes open-heart surgery, haemodialysis when a patient's kidneys fail, and intravenous (IV) nutritional and electrolyte (for example, potassium and sodium) support when a patient is unable to eat or drink.

The ICU or ICCU, in hospitals are designed to offer the advantages of a low concentration of the equipment and resources needed to take care of critically ill or seriously injured patients. Hence, there should be specially trained personnel who have physical resources, supplementing their skills and abilities, to handle emergencies. As a part of this concept a modular controller is designed to carry out monitoring of some usual biological parameters.

In the current scenario the patient monitoring system using GSM or GPS is introduced. This system are designed to monitor the patient in network from remote location. This system can be designed using different
wireless technology and sensors. A devise which locate alive people in war area is also introduced by using pulse rate sensor.

III. CONSTRUCTION & DESIGN

The required components used in this project are Power supply, Heart beat sensor and temperature sensor, Microcontroller (ATMEGA16), GSM modem, GPS receiver, Liquid Crystal Display (16*2).

**Block Diagram:**

![Block Diagram](image)

Microcontroller is heart of system. Microcontroller ATMEGA16 is programmed to get input from sensors and send output as SMS by accessing GSM modem using AT commands. Heart beat detector gives digital output to microcontroller at input capture pin. Temperature sensor LM35 gives analog voltage to microcontroller, whose voltage is linearly proportional to centigrade value and GPS receiver gives NEMA format serial data to microcontroller. On output side LCD0 is connected to PORT C of microcontroller, heart beat monitoring is continuously displayed on LCD with beats per minute readings.

As soon as patient inserts his finger inside heart beat detector, microcontroller counter is started for 15 second and heart beat counted in these 15 second multiply by four gives beats per minutes value (BPM)

\[
BPM = \text{BEATin15sec} \times 4
\]

If beats per minute is greater than preset value (say 120 BPM) then temperature value is calculated by equation given below

\[
\text{TEMPRATURE}(°C) = \text{Vout} \times (100°C/V)
\]

Now serial data will be taken from GPS receiver and as per format longitude and latitude are extracted from serial data. Microcontroller now generates SMS using AT commands, AT commands access GSM modem with RS232 pin interface. This message contains information of Beats per minute, Temperature, Latitude, Longitude.

Patient physiological data will be sent to a number of doctors or ambulance centre, which will track patient position if patient beats are abnormal in cases.

A. Heartbeat Sensor:

Features of heartbeat sensor includes Heat beat indication by LED, It has instant output digital signal for directly, It always connected to microcontroller, Compact in Size, It’s working voltage is +5V DC.

B. Temperature Sensor LM35:

Features of temperature sensor include It calibrated directly in ° Celsius (Centigrade) and 0.5°C accuracy guarantee able (at +25°C) with full -55° to +150°C range, it is also suitable for remote applications, Low cost due to wafer-level trimming, Operating range from 4 to 30 volts with low self-heating, it’s temperature 0.08°C in air.

C. ATMEGA16 Microcontroller:

The ATmega16 is a low-power CMOS 8-bit microcontroller based on the AVR enhanced RISC architecture. By executing powerful instructions in a single clock cycle, the ATmega16 achieves throughputs approaching 1 MIPS per MHz allowing the system designed to optimize power consumption versus processing speed. The basic architecture of ATMEGA16 consists of the following features like High-performance, Low-power AVR8-bit Microcontroller, Nonvolatile Program and Data Memories, 16K Bytes of In-System Self-Programmable Flash, Advanced RISC Architecture, 131 Powerful Instructions – Most Single-clock Cycle Execution, 32 x 8 General Purpose Working Registers, Fully Static Operation, Endurance: 10,000 Write/Erase Cycles, Optional Boot Code Section with Independent Lock Bits, In system programming by On-chip Boot program, True Read-While-Write operation, 512 Bytes EEPROM, Endurance: 100,000 Write/Erase cycles 1K byte internal SRAM, programming lock for Software Security, JTAG (IEEE std.
1149.1 Compliant) Interface, Boundary-scan capabilities according to the JTAG Standard, Extensive On-chip debug support, programming of Flash, EEPROM, Fuses, and Lock Bits through the JTAG Interface. The another special Features of Microcontroller are Power-on Reset and Programmable Brown-out Detection, Internal Calibrated RC Oscillator/I/O and Packages, 32 Programmable I/O Lines, 40-pin PDIP, 44-lead TQFP, and 44-pad MLF. Operating Voltages: 4.5 - 5.5V for ATmega16, Speed Grades0 - 16 MHz for ATmega16External and Internal Interrupt Sources

D. Power Supply:
AC – DC Power Adaptor with DC Voltage: 12V andDC Current: 1A ratings.

E. GPS Receiver:
Features of GPS Receiver are High sensitivity -160dbm, LED indicating data output, Low power consumption, GPS L1 C/A Code, Supports NMEA0183 V 3.01 data protocol, Real time navigation for location based services, Works from +5V DC signal and outputs 9600 bps serial data. Magnetic base active antenna with 3 meter wire length for vehicle rooftop installation.

F. GSM Modem:
This GSM modem is a highly flexible plug and play quad band GSM modem for direct and easy integration to RS232. Supports features like Voice, Data/Fax, SMS, GPRS and integrated TCP/IP stack. Purpose of using GSM modem is to increase range of transmission of patient data from anywhere to hospital usually required. Feature of this modem are Quad Band GSM/GPRS 850/900/1800/1900 MHz, Control via AT commands (GSM 07.07, 07.05 and enhanced AT commands).

G. Liquid Crystal Display:
A liquid crystal display (LCD) is a thin, flat display device made up of any number of color or monochrome pixels arrayed in front of a light source or reflector. It uses very small amounts of electric power, and is therefore suitable for use in battery powered electronic devices. Each pixel consists of a column of liquid crystal molecules suspended between two transparent electrodes, and two polarizing filters, the axes of polarity of which are perpendicular to each other. The liquid crystal twists the polarization of light entering one filter to allow it to pass through the other.

More microcontroller devices are using ‘smart LCD’ display to output visual information. LCD displays designed are inexpensive, easy to use, and it is even possible to produce a read out using the 8*80 pixels of the display. LCD displays have a standard ASCII set of characters plus Japanese, Greek and mathematical symbols. For an 8-bit data bus, the display requires a +5V supply +11 I/O lines.

1. IMPLEMENTATION

A. Wearable Sensors
The wearable sensors provide four functionalities: vital signs monitoring location-tracking, medical record storage, and triage status tracking. We integrated two types of non-invasive vital signs sensors – a pulse oximeter and a blood pressure sensor. The pulse oximeter attaches to the patient’s finger and measures heart rate (HR) and blood oxygenation level (SpO2). A cuff pressure sensor on the patient’s upper arm measures systolic and diastolic blood pressure. We also integrated two types of location sensing capabilities – a GPS to provide geolocation, and indoor location detection system to provide location where the GPS signal cannot be reached. The GPS sensor allows medics to track patients who are outdoors, e.g. at the scene of the emergency, with accuracy of 3 meters (CEP). The indoor location system, based on the Mote Track project developed at Harvard University, requires the installation of location beacons. Indoor location beacons are being installed at a designated auxiliary care center near Washington DC. Patients are admitted to an auxiliary care center if nearby hospitals have reached their occupancy capacities and cannot admit more patients. At an auxiliary care center, which can often be short on staff and overfilled with patients, the patients vital signs will continue to be monitored by our system. Johns Hopkins University Applied Physics Laboratory medics quickly locate a specific patient whose conditions have deteriorated. With all the peripheral devices turned on, the pulse and oxygenation reported every second, the GPS location reported every 5 minutes, and the blood pressure reported every 15 minutes, and the battery lifetime of the overall system is approximately 6 hours. The blood pressure sensor is the most power hungry peripheral, and when it is not used, the battery life of the overall device increases to 1-2 days.

B. Vital Sign Monitor Algorithm
Software on the tablet device receives real-time patient data from the mote and processes them to detect anomalies. If the patient has a medical record that has been previously entered, information from the medical record is used in the alert detection algorithm. Table shows a partial list of physiological conditions that cause alerts. The algorithm uses additional information such as patient age and height to adjust its thresholds. If additional information is not available, the algorithm uses a set of default values.

C. Web Portal
An effective emergency response information system should support the need for multiple parties to share information about patients’ status and locations. Our web based
information portal allows different types of users to access the patient information in real-time. When a user logs in, the information displayed to that particular user is managed by group-level permissions.

IV. CONCLUSIONS

We have designed and developed a real-time patient monitoring system that integrates vital signs sensors, location sensors, ad-hoc networking, electronic patient records, and web portal technology to allow remote monitoring of patient. This system provides an exact condition of patient situated in remote area. We can monitor multiple patients in a hospital using single monitor in a doctor’s cabin. Limitations of the previously used patient monitoring systems include the measurement of each parameter using separate devices and hence resulting in overcrowding of devices around the patient which also disturbs the situation of patient. The patient monitoring system developed in this project overcomes this disadvantage by the construction of an integrated circuit comprising of all the parameter measuring devices and organized acquisition of data and their meaningful display.

REFERENCES

[5] International Journal of Electronics and Communication Engineering & Technology (IJECEIT), ISSN 0976 – 6464(Print), ISSN 0976 – 6472(Online) Volume 4, Issue 1, January-February (2013), © IAEME.
“Demonstration of Pulsed X-ray Machine Radiography as an Alternative to Industry Radiography Cameras, Demonstration Pilot Project”

Vitthal P. Shirsat#, Thaneshwar R. Patle*2, Mukta R. Deshmukh#3

#Students of Mechanical Engg. Dept. From Dr. Bhausaheb Nandurkar college of Engg. & Tech. Yavatmal in Amravati University

Abstract—
In this Paper Presentation we conclude the Conventional X-ray machining related to Biomedical and mechanical research. This will helps to new era of technological aspects of the biomedical industries and fulfill the lack of improved sources such as heating of X-ray machines, more efficiently uses of latest technologies that will modify our lifestyle. This research has been done according to the fundamentals of Biomedical & Mechanical Engineering fields. And helps to improve the patient’s condition according to the related field and studies etc. This will also helps to reduce the power consumption of the whole unit. And helps to increase the machine life and overall life cycle for the commercial purpose. All the latest studies regarding to this topic is as given below.

XVIII. INTRODUCTION
The Radiation Protection Division of the Environmental Protection Agency (EPA) is dedicated to minimizing incidences of lost radioactive sources that enter into consumer metal supplies and the public domain. Industrial devices and consumer products containing radioactive sources routinely fall out of regulatory control. Once out of regulatory control, these devices and products may be subjected to harsh conditions capable of producing a breached source, with the potential of harmful exposure incidents and significant economic impacts to industry. Providing alternative technologies for devices and products which utilize radioactive sources is one approach to minimize lost source incidences. The current focus of EPA’s efforts in this regard is to conduct those studies and assessments necessary to support the implementation of such alternative technologies in industrial practices – alternatives that are technologically and economically advantageous. The approach suggested by Southwest Research Institute® (SwRI®) is to identify an industrial sector that routinely uses isotopic radiation sources and to demonstrate that an alternative technology to isotopic sources can provide equivalent capability. One industrial sector that regularly uses isotopic sources to perform radiography of pipeline welds is the pipeline industry. The industry uses Co60, Cs137, and Ir192 which have gamma ray energy lines of 1.17 and 1.33 MeV, 0.66 MeV, and 0.31, 0.47, and 0.60 MeV. Ir192 is perhaps the most often used source for pipeline welds because the pipe wall thicknesses usually range between 0.25 and 0.4 inches. Ir192 has a half-life of 74.3 days. Sources are usually purchased with an activity of approximately 100 curies. The radiography conducted is usually double wall for detecting cracks, inclusions, and porosity in the welds as illustrated in Figure 1. To verify the quality of the radiography, the code that regulates the radiographic inspection usually calls for a “penetrameter” or “image quality indicator (IQI)” and the image sensitivity required. In these radiographs an IQI was used, and the quality requirement was that all the wires had to be detected.

XIX. TECHNICAL APPROACH
The goal of this project was to demonstrate a radiography technology for inspection of pipe welds that does not require the use of isotopic sources. The technical approach to be followed included (1) developing procedures for inspection of schedule 40 pipe in the range of 3 to 16 inches in diameter, (2) producing radiographs with both an Ir192 source and a pulsed, battery operated, portable X-ray source with a peak X-ray energy of 270 kV and (3) comparing the results obtained as well as the operational issues associated with using the X-ray source compared to the isotopic source.

Isotopic Source and Pulsed X-ray Source:
Isotopic sources (called “pills”) are very small, often on the order of approximately ¼ inch diameter by ¾ inch long. The pill is usually contained in a shielded housing usually called a “camera.” Although these sources are highly regulated, because they are so small they can easily be inadvertently or intentionally removed from the regulatory information stream. Isotopic source technology has a number of advantages over existing large X-ray sources. For example, the source technology employs a very compact geometrical envelop and does not require...
any electrical power. Conventional x-ray sources, on the other hand, require 220V power and room for a cooling system (often water based). In addition, Ir192 provides very good radiographs, and this source has been used for many decades so that the knowledge base on its use is well accepted. However, recent advancements have been made in pulsed x-ray sources that operate using 14.4-volt battery power and have a geometrical envelop similar to the isotopic source shielded housing.

XX. DISCUSSION

The approach suggested by SwRI was to demonstrate the capabilities of a 270KVP pulsed, battery powered x-ray unit and to compare the double wall pipe radiographs generated using the pulsed x-ray source with the real-time imaging device to radiographs generated with Ir192 and film. The demonstration application was double wall radiography (where the source is placed on one side of the pipe and the film or imager is on the other side of the pipe) for a variety of pipe welds (ranging in diameter from 4 to 16 inches) as illustrated in Figure 4. The pipes were fabricated with intentionally placed defects. The welders intentionally used poor welding techniques to generate regions of lack of fusion, lack of penetration, porosity, and slag. The intent was to develop a number of regions where natural defects occurred as well as a number of regions where there were few or no defects. Examples of the pipes and the types of defects are illustrated in Figure 1.

Figure 1. Illustration of source size and film/detector set used for the isotopic and pulsed x-ray sources

Impact to End-Users:

Issues to be addressed included differences in the procedures in terms of time, set up, personnel required, source cost and labor/cost associated with following regulations, ease of use, and a discussion of the likelihood of successfully transferring this technology to industry. In terms of information obtained from the isotopic radiograph and the pulsed x-ray source used in conjunction with the Vidisco real-time imaging system for this pipe diameter and thickness range, the two were basically identical. In terms of procedure development, the pulsed x-ray/Vidisco procedure development was faster because images could be obtained within a few minutes of the actual exposure as compared to at least 30 minutes needed for film development to develop the isotopic procedure. In terms of geometrical issues, the vendor still preferred the isotopic source because it is very small and can be taped directly onto the pipe. In terms of actual radiation exposure time, for the isotopic source (depending on wall thickness) the exposure time was on the order of 60 seconds per shot and for the pulsed x-ray source, the exposure time for the same weld was approximately 3 seconds. Labor cost using the isotopic source is approximately double the cost associated with using the pulsed x-ray source because regulations require that two radiographers must be present when using an isotopic source while only one radiographer is required when using an x-ray machine. In terms of ease of use, the pulsed x-ray source/real-time imager is very similar to the isotopic source radiography. In terms of the cost of using the isotopic source film technology versus pulsed x-ray source/real-time imaging technology, the information contained provides comparisons between the two technologies by the vendor used in the pilot demonstration.

COMMERCIALIZATION PLAN:

To effectively commercialize this technology, an inspection company that performs pipeline weld inspections, such as All American Inspections, must be convinced that this technology is useful. To that end, All American is an integral part of the team. Secondly, SwRI and All American will present the results obtained from this pilot demonstration at an ASNT (American Society of Nondestructive Testing) conference and an API (American Petroleum Institute) meeting. These presentations will provide a good opportunity to showcase the technology and to share information with companies that usually conduct pipeline inspections using isotopic sources. These presentations will most likely occur at the 2007 ASNT Fall Conference and the 2007 API Conference. Aerospace application involved in the maintenance and repair of aerospace structures could be a viable commercial area. Perhaps evaluating the system at the FAA NDI validation center would provide some credibility to the system and its capabilities. As a provider of services, All American would be able to market the system easily. Durability of the system and duty cycle would need to be evaluated, but the potential is there.

XXI. CONCLUSIONS

Based upon the work conducted to date on this project, the following conclusions have been reached.

(1) For double wall pipe radiography (which is the requirement for field pipeline weld joint inspection), isotopic radiography and pulsed x-ray with real-time imaging capability provide results that
meet the code requirements. The nominal code requires that ASME IQI B wire (all wires) can be detected.

(2) This system is excellent for the intended DOT pipeline inspection, providing the adequate sensitivities are achieved. Because of its portability and reduced exposure time this seems to be a “great fit”.

(3) Contacting ASTM and approaching their radiographic committee in an effort to address this type of system specifically would be a vehicle to encourage its use industry wide.

References

[38] Shibazaki, aaaaaan. A & Lamb, F. Astrophy .J.346,808-822
Cloud Computing – A Review

Prof. Hemant P. Moodliar
Associate Prof. & Head, Department of Information Technology
Dr. BhausahebNandurkar Col. of Engg.& Tech.,Yavatmal

Miss. Tejashri K. Lakhpati
Student, Dept. of IT
DBNCOET, Yavatmal

Abstract- Cloud Computing has gained popularity in both research and industrial communities. Cloud users can acquire computing resources on a need basis, achieving on demand scalability; Cloud providers can maximize resource utilizations of data centres, increasing their return on investments. While Cloud systems are usually hosted in large data centres and are centrally managed, other types of Cloud architectures can be imagined. Cloud computing platform is a set of scalable large-scale data server clusters, it provides computing and storage services to customers. The basic study about the architecture of current cloud computing system shows that it’s a central structured one; i.e. All the data nodes are indexed by a master server, but when the number requests increases it may become bottle neck of the system. This research paper is about a cloud storage architecture based on P2P with fault tolerance. Here the central entity is removed and all servers are interconnected to form a ring structure. When one of the servers fails, the work will be taken over by any of the best performing servers. The confidentiality and integrity of data passed in between the servers is also maintained using MAC algorithm. Cloud computing is set of resources and services offered through the Internet. Cloud services are delivered from data centres located throughout the world. Cloud computing facilitates its consumers by providing virtual resources via internet. General example of cloud services is Google apps, provided by Google and Microsoft SharePoint. The rapid growth in field of “cloud computing” also increases severe security concerns. Security has remained a constant issue for Open Systems and internet, when we are talking about security cloud really suffers. How the end users of cloud computing know that their information is not having any availability and security issues? Every one poses, Is their information secure? This study aims to identify the most vulnerable security threats in cloud computing, which will enable both end users and vendors to know about the key security threats associated with cloud computing. Our work will enable researchers and security professionals to know about users and vendors concerns and critical analysis about the different security models and tools proposed.

Index Term: cloud computing, P2P, storage, cloud computing security, secure cloud computing.
INTRODUCTION

“Cloud computing” simply means Internet computing, generally the internet is seen as collection of clouds; thus the word cloud computing can be defined as utilizing the internet to provide technology enabled services to the people and organizations. Cloud computing enables consumers to access resources online through the internet, from anywhere at any time without worrying about technical/physical management and maintenance issues of the original resources. Besides, Resources of cloud computing are dynamic and scalable. Cloud computing is independent computing it is totally different from grid and utility computing. Google Apps is the paramount example of Cloud computing, it enables to access services via the browser and deployed on millions of machine over the Internet. Resources are accessible from the cloud at any time and from any place across the globe using the internet. Cloud computing is cheaper than other computing models; zero maintenance cost is involved since the service provider is responsible for the availability of services and clients are free from maintenance and management problems of the resource machines.

The key features of the cloud computing are:
1. The cloud computing design is done in such a way that the resources present in it will be available from anywhere at anytime.
2. Since replication of data is done in cloud computing, the resources are available even during hardware failure.
3. Cloud computing provides greater speed in its operation.
4. The on-demand application deployment increases the resource utilization to a large extend.
5. Low cost servers are available for storage and services.

Cloud computing system [2] has the capability to hold heavy load situations without much hardware support. It makes use of the virtualization concept. For client based transactions its better to store the data in cloud. Google File System [4] is a scalable distributed file system for large distributed data-intensive applications. It provides fault tolerance while running on inexpensive commodity hardware, and it delivers high aggregate performance to a large number of clients. The file system has successfully met different storage needs. It is widely deployed within Google as the storage platform for the generation and processing of data used by our service as well as research and development efforts that require large data sets. The strength of their work is the proposed secure provenance system and limitation of their work is that their proposed scheme is difficult to implement as it is based on complex mathematical model which is very difficult to understand.

La’Quata Sumter et al. [2] says: The rise in the scope of cloud computing has brought fear about the Internet Security and the threat of security in cloud computing is continuously increasing.

Mladen [3] states that —Cloud computing is a recent field, which came into existence after Years of research in networking and different types of computing.

LITERATURE SURVEY

Due to feature, cloud computing is also known as utility computing, or “IT on demand”. Scalability is key attribute of cloud computing and is achieved through server virtualization. This fresh, web-based generation of computing uses remote servers placed in extremely safe and secure data centres for storage of data and management, so organizations do not need to pay for and look after their internal IT solutions.

cloud applications use large data centres and powerful servers that host web applications and web services. Anyone with a suitable Internet connection and a standard browser can access a cloud application.
EXISTING SYSTEM

Now days a single server has the capability to handle the multiple requests from the user. But the server has to process the all the requests from the user parallel, so it will lead to a hike in the processing time of the servers. This may leads to loss of data and packets may be delayed and corrupted. On doing this the server cannot process the query from the user in a proper manner. So the processing time gets increased. It may leads to traffic and congestion. To overcome these problems we are going for the concept called “cloud computing”. In this cloud computing we are going to implement the chunk server to avoid these problems. Google cloud computing infrastructure has four systems which are independent of and closely linked to each other. They are Google File System for distributed file storage, Map Reduce program model for parallel Google applications, Chubby for distributed lock mechanism and Big Table for Google large-scale distributed database.

SYSTEM MODEL

We consider a large set of networked nodes which can be owned by different individuals or organizations. Each node includes a processor, RAM, storage space and network connectivity; we do not require that all nodes are homogeneous. Users of this system (which in general are the owners of the nodes) share the resources (CPUs, memory, disks) cooperatively. To do so, they install a software daemon on each node (Figure 2) which takes care of maintaining cohesion and gracefully handle churn; in fact nodes are not required to be reliable, so they can join or leave the system at any time. The software daemon has two separate interfaces: a user interface, through which users can inject requests into the system, and a node-to-node interface which is used to communicate with other peers.

P2PCS ARCHITECTURE

In this section we give a high-level description of the P2PCS architecture. We focus on algorithmic and protocol issues; additional details will be given in Section 5.
As already discussed, P2PCS is implemented as a collection of identical interacting processes, each one running on a separate host. Each process is made of several software modules that are roughly organized according to the layered structure shown in Figure 3. The Peer Sampling Service (PSS) [15] aims at providing each node with a list of peers to exchange messages with; this is achieved by maintaining an unstructured overlay over the set of peers.

Figure 4: Creating subclouds in P2PCS

The storage policies of P2P say that:
1. The cost to the P2P system will be lower if one allocates large files to unreliable peers, and assigns smaller files to reliable peers.
2. Unreliable peers should be allowed to distribute less, and reliable peers should be allowed to distribute more.

**PROPOSED SYSTEM**

New cloud storage architecture based on P2P and designs a prototype system. A cluster consists of a single database and multiple chunk servers and is accessed by multiple clients. Chunk servers store chunks on local disks and read or write chunk data specified by a chunk handle and byte range. The database maintains all file system metadata. When a client wants to visit some data on a chunk server, it will first send a request, and the database then directs with the corresponding chunk handle and locations of the replicas. Hence the processing loads on servers are balanced.

The architecture consists of mainly 3 modules:

**FUTURE WORK**

Cloud computing is not fully mature and still lot needs to be explored. After our current work we are claiming that security is the most important threat to both the users and the vendors of cloud computing. Vendors, Researchers and IT security professionals are working on security issues.
associated with cloud computing. Different models and tools have been proposed but still nothing fruitful found. While doing research on security issues of cloud computing we came to know that there are no security standards available for secure cloud computing. In our future work we will work on security standards for secure cloud computing. The future work of this proposed system could to modify the system performance by reducing the number servers present in the network.

CONCLUSION

In this study different security and privacy related research papers were studied briefly. Cloud services are used by both larger and smaller scale organizations. Advantages of Cloud computing are huge. But it's a global phenomenon that everything in this world has advantages as well as disadvantages. We propose a cloud computing architecture based on P2P which provide a pure distributed data storage environment without any central entity for controlling the whole processing. The advantage of this is architecture is that it prevents the bottleneck problem that arises in most of the client server communications. The proposed system does its operation based on the performance of the system. It does the monitoring operation to find out the best chunk servers within the P2P network. It does this operation in order to perform efficient resource utilization and load balancing of the servers. In this paper we described the architecture and prototype implementation of P2PCS, a fully distributed IaaS Cloudsystem.

REFERENCES


[34] Antony Rowstron and Peter Druschel, “Pastry: Scalable, decentralized object location and routing for large-scale peer-to-peer systems”, IFIP/ACM International Conference on Distributed Systems Platforms.

Neuromorphic Brain Chips: Microchips That Imitate the Brain

Snehal D. Bajare¹, Mohan H. Faltankar², Imran S. Khan³

¹UG student Department of Electronics & Telecommunication, DBNCOET, Yavatmal, SGBAU, Amravati, (ms) India
²UG student Department of Electronics & Telecommunication, DBNCOET, Yavatmal, SGBAU, Amravati, (ms) India
³Assistant Professor, Department of Electronics & Telecommunication, DBNCOET, Yavatmal, SGBAU, Amravati, (ms) India

Abstract—This document gives formatting instructions for individuals with severe disabilities facing challenges performing normal everyday tasks. No computer works as efficiently as the human brain – so much so that building an artificial brain is the goal of many scientists. The technology used is neurotechnology with BCI (Brain Computer Interface). It has great results in the upcoming years. The study of the human brain is, obviously, the most complicated area of research. It will be several years before we see a practical application of the technology we’ve discussed. Let’s hope such technologies will be used for restoring the prosperity and peace of the world and not to give the world a devastating end.

XXII. INTRODUCTION

Thousands of people around the world suffer from paralysis, rendering them dependent on others to perform even the most basic tasks. But that could change, thanks to the latest achievements in the Brain-Computer Interface, which could help them regain a portion of their lost independence. A four-millimetre square silicon chip studded with 100 hair-thin microelectrodes, is embedded in the primary motor cortex - the region of the Brain responsible for controlling movement. When somebody thinks “move cursor up and left,” his cortical neurons fire in a distinctive pattern; the signal is transmitted through the pedestal plug attached to the skull. The signal travels to an amplifier where it is converted to optical data and bounced by fibre-optic cable to a computer. Brain Gate learns to associate patterns of brain activity with particular imagined movements—up, down, left, right—and to connect those movements to a cursor. Using BCI advancement wirelessly, which is programmed and directly implanted inside the brain called neuromorphic brain chip which can allow a paralysed person on wheelchair to stand on their own feet, deaf to hear, blind to see.

XXIII. EARLY FINDINGS

Neurotechnology the application of electronic engineering to the human nervous system, has now reached a level of commercial and scientific maturity that will produce enormous benefits to patients and profits to investors. Hundreds of thousands of people have already been helped by neurostimulation products that restore hearing to deaf people, movement to individuals with paralysis, and relief to those with chronic pain or neurological disorders. Please join us as we monitor the growth of this new industry and track the progress from medical technology to commercial products. Neurotechnology is any technology that has a fundamental influence on how people understand the brain and various aspects of consciousness, thought, and higher order activities in the brain. It also includes technologies that are designed to improve and repair brain function and allow researchers and clinicians to visualize the brain. The field of neurotechnology has been around for nearly half a century but has only reached maturity in the last twenty years. The advent of brain-imaging revolutionized the field, allowing researchers to directly monitor the brain’s activities during experiments. Neurodevices are any devices used to monitor or regulate brain activity. The most common neurodevices are deep brain stimulators (DBS) that are used to give electrical stimulation to areas stricken by inactivity. Neurotechnology is the application of medical technology, devices and treatments to improve functions throughout many parts of the body. Hundreds of thousands of people with neurological or psychiatric disabilities have already been helped by neurotechnology. For instance, cochlear implants help restore hearing to the deaf, neural prosthesis provide movement to the paralyzed, allow locked-in to communicate and spine stimulators give relief to those with chronic pain. These are just a few examples of the many medical technologies for people with disabilities that are available today.

III. BRAIN-COMPUTER INTERFACE

A brain–computer interface (BCI), often called a mind–machine interface (MMI), or sometimes called a direct neural interface or a brain–machine interface (BMI), is a direct communication pathway between the brain and an external device. Brain Device Moves Objects by Thought. BCI is a new technology which can let human to control electronic devices without lifting a finger. It simply reads brain activity. Research on BCIs began in the 1970s at the University of California Los Angeles (UCLA) under a grant from...
the National Science Foundation, followed by a contract from DARPA. The field of BCI research and development has since focused primarily on neuroprosthetics applications that aim at restoring damaged hearing, sight and movement. Thanks to the remarkable cortical plasticity of the brain, signals from implanted prostheses can, after adaptation, be handled by the brain like natural sensor or effector channels. Neuroprosthetics is an area of neuroscience concerned with neural prostheses. That is, using artificial devices to replace the function of impaired nervous systems and brain related problems, or of sensory organs.

**Fig 2. BCI Architecture**

**BCI is basically of three types:**
1. Non-invasion
2. Partial invasion
3. Invasion

**III.1 Non-invasion BCI**

This technology observes the activity of the brain by just fixing the device on the scalp of the brain. Although they are easy to wear, non-invasive implants produce poor signal resolution because the skull dampens signals, dispersing and blurring the electromagnetic waves created by the neurons. Although the waves can still be detected it is more difficult to determine the area of the brain that created them or the actions of individual neurons. Electroencephalography (EEG) is the most studied potential non-invasive interface, mainly due to its fine temporal resolution, ease of use, portability and low set-up cost. in experiments beginning in the mid-1990s, Niels Birbaumer at the University of Tübingen in Germany trained severely paralysed people to self-regulate the slow cortical potentials in their EEG to such an extent that these signals could be used as a binary signal to control a computer cursor (Birbaumer had earlier trained epileptics to prevent impending fits by controlling this low voltage wave.) The experiment saw ten patients trained to move a computer cursor by controlling their brainwaves. The process was slow, requiring more than an hour for patients to write 100 characters with the cursor, while training often took many months.

**III.2 Partial invasion**

Partially invasive BCI devices are implanted inside the skull but rest outside the brain rather than within the grey matter. They produce better resolution signals than non-invasive BCIs. Electrocorticography (ECoG) measures the electrical activity of the brain taken from beneath the skull in a similar way to non-invasive electroencephalography but the electrodes are embedded in a thin plastic pad that is placed above the cortex, beneath the dura mater. This research indicates that control is rapid, requires minimal training, and may be an ideal tradeoff with regards to signal fidelity and level of invasiveness.

ECoG is a very promising intermediate BCI modality because it has higher spatial resolution, better signal-to-noise ratio, wider frequency range, and less training requirements than scalp-recorded EEG, and at the same time has lower technical difficulty, lower clinical risk, and probably superior long-term stability than intracortical single-neuron recording. This feature profile and recent evidence of the high level of control with minimal training requirements shows potential for real world application for people with motor disabilities. The EEG technology's susceptibility to noise, another substantial barrier to using EEG as a brain–computer interface is the extensive training required before users can work the technology.

**III.3 Invasion**

Invasive BCI research has targeted repairing damaged sight and providing new functionality for people with paralysis. Invasive BCIs are implanted directly into the grey matter of the brain during neurosurgery. Because they lie in the grey matter, invasive devices produce the highest quality signals of BCI devices. So here the neurosurgery has to be used.

**Fig 3 Brain Computer Interface**
IV. THE CHIP

Output brain chip (or Brain-Computer Interface) technology functions by monitoring the electrical activity of a small section of the brain, mathematically interpreting the data, and relaying the new information to an action-performing device. Internal sensing devices, or brain chips, can listen much more closely. Placed into the brain itself, the electrode arrays of these chips come into direct contact with live neurons, and so can sense single neuron impulses. Sensing devices for brain-computer interfaces come in two basic flavors, those internal to the skull, and those that are external. External devices sense electrical activity through the layers of skin, blood, connective tissue, and bone that separate them from their queries. Current methods of direct neuron sensing being tested in humans use arrays of as many as 100 micro-electrodes, recording the electrical activities of up to 96 different neurons or small groups of neurons at a time.

An ideal micro-electrode array for nerve cell recording needs to be small, sturdy, and harmless. It should fit into the brain without squishing anything, should not wear out or move around in spite of constant vibration and jarring head movements, and should not interact with the cells around it in any way other than to record electrical information. Researchers around the world, mostly in academic settings, are developing a variety of designs to meet these requirements for a safe and effective device.

V. WORKING OF BRAIN CHIPS

As we read this article and think, the nerve cells in our brains are receiving, interpreting, and responding to a bombardment of information. The light emitted from every pixel on this page causes a different bunch of sensory neurons in our eyes to light up in a flurry of electrical activity, conveying the text and images to the brain, while as many as 100 billion neurons network the electrical signals, forming our thoughts and directing our actions. Neurons conduct information as electrical impulses, internal fluctuations in voltage. Normally they do this to communicate amongst themselves, however any electrical device can potentially ‘speak’ the language of a neuron if only it could listen closely enough. It is by listening closely that brain-computer interface technology is able to interpret the languages of our neurons. Placed into the brain itself, the electrode arrays of these chips come into direct contact with live neurons, and so can sense single neuron impulses. Current methods of direct neuron sensing being tested in humans use arrays of as many as 100 micro-electrodes, recording the electrical activities of up to 96 different neurons or small groups of neurons at a time.

VI. THE BRAIN

The brain acts as the command and control center for the human body. Its ability to integrate numerous signals to and from various sources underlies the complex behavior of humans. The brain controls basic functions like breathing, tasting, and moving but in addition, it is the basis for personality, it generates emotions and it is the center for consciousness.

The forebrain, midbrain, hindbrain, and spinal cord form the central nervous system (CNS), which is one of two great divisions of the nervous system as a whole. The brain is protected by the skull, while the spinal cord, which is about 17 inches (43 cm) long, is protected by the vertebral column.

Fig.4. The Brain

The nervous system consists of two parts, the Central Nervous System (CNS) and the Peripheral Nervous System (PNS). The brain and the spinal cord constitute the CNS and the motor and sensory nerves that lie outside the CNS constitute the PNS. The brain, weighing approximately 3 pounds, is divided into two hemispheres which are each separated into four lobes – frontal, parietal, temporal, and occipital. The frontal lobe controls reasoning, planning, speech, movement (motor cortex), emotions, and problem-solving. The parietal lobe is concerned with perception of stimuli related to touch, pressure, temperature, and pain. The temporal lobe is responsible for perception and recognition of auditory stimuli and memory while the occipital lobe focuses on vision. Measuring about 18 inches in length and weighing about 35 grams, the spinal cord is the main pathway to connect passage of information between the brain and the PNS.

The primary motor cortex (located in the frontal lobe) is the main source of voluntary movement signals. This area is divided into specific regions to control distinct parts of the body. From studying the brain, it appears as though the control of the body parts is highly distributed within the region meaning that, for example, the neurons in the arm region as well as the neurons surrounding the arm region are capable of controlling signals for the arm. As a result, the primary motor cortex is an ideal site for the BCI because of this distribution.

VII. EXPERIMENT

"Animal studies show that the motor cortex does adapt as the animal learns something new," he said. "It’s like learning to use a tennis racket until it becomes an Extension of your own arm." Will normally healthy people ever be able to use a brain chip to enhance their memory, strength, vision or other functions? Hatsopoulos says that’s more science fiction right now and...
would raise ethical issues because the procedure involves surgery to implant the device on the brain.

Figure 5. Experiment
Department of Physiology and Biophysics, University of Washington School of Medicine in Seattle, showed for the first time that monkeys could learn to control the robots only by their thought.

VII. Mathew Nagle

With a tiny electronic chip implanted in the Motor cortex of his brain, a 25-year-old man paralyzed from the neck down for five years has learned to use his thoughts to operate a computer, turn on a TV set, open email, play a video game and manipulate a robotic arm. The chip that’s implanted in their brains is about the size of a baby aspirin. It has 100 electrodes, each thinner than a human hair, that pick up the electronic chatter from between 30 to 60 neurons in the motor cortex, which normally controls arm movement. The device, called the Brain Gate Neural Interface System, is produced by Cyberkinetics of Foxborough, Mass., which was originally established by Donoghue and others. The patient imagines moving his arm to activate the neurons. The chip registers this activity, which is then converted into a program for controlling a computer cursor, TV, e-mail and other devices, and patients quickly learn how to adjust their thought processes to control the different systems. It only takes minutes, for example, for a patient to imagine moving his arm to track a moving cursor on a computer screen, and then to be able to move the cursor with his own thoughts.

VIII. Achievement

Jennifer was confined to a wheelchair (left) for seven years by dystonia, a condition that causes uncontrollable muscle spasms. Now (right) she walks without assistance, thanks to battery-powered electrodes that were implanted in her brain when she was 13—and to surgeries that then repaired her twisted muscles and lengthened her tendons.

Fig. 7 Development of Jennifer

On My Feet Again is the inspiring and intellectually stimulating story of how a determined and resourceful young woman overcame many of the obstacles that came her way after being paralyzed in a snowboarding accident. Although told she would never get out of her wheelchair, Jennifer refused to accept that fate and sought out experimental new technologies for people with spinal cord injuries. She became a participant in a clinical trial of a neuroprosthetic system that enables her to stand up out of her wheelchair and move around on her own two feet using her own muscles, which are stimulated by 24 surgically implanted electrodes. Along the way, she helped advance the technology by working with the research team to test new features, push the limits of her strength and endurance, and prove the viability of this new therapy for thousands of other potential recipients—including injured soldiers returning from war. In the book, Jennifer describes how the neurotechnology system she helped to perfect became a key factor in her conditioning and competition at the 2012 Paralympics Games in London, where she brought home a silver medal in sailing for Team USA.

IX. Advantages

The future may well involve the reality of science fiction's cyborg, persons who have developed some intimate and occasionally necessary relationship with a machine. It is likely that implantable computer chips acting as sensors, or actuators, may soon assist not only failing memory, but even
bestow fluency in a new language, or enable “recognition” of previously unmet individuals. The progress already made in therapeutic devices, in prosthetics and in computer science indicates that it may well be feasible to develop direct interfaces between the brain and computers.

Computer scientists predict that within the next twenty years neural interfaces will be designed that will not only increase the dynamic range of senses, but will also enhance memory and enable "cyberthink" - invisible communication with others. This technology will facilitate consistent and constant access to information when and where it is needed.

The linkage of smaller, lighter, and more powerful computer systems with radio technologies will enable users to access information and communicate anywhere or anytime. Through miniaturization of components, systems have been generated that are wearable and nearly invisible, so that individuals, supported by a personal information structure, can move about and interact freely, as well as, through networking, share experiences with others. The wearable computer project envisions users accessing the Remembrance Agent of a large communally based data source.

- It will increase the dynamic ranging of senses.
- Giving light to blind and giving paralyzed patients full mental control of limbs.
- Rescue missions (remote controlled rat).
- It will enable “cyberthink”.
- It will enable consistent and constant access to information where and when it is needed.
- The advantage of implants is that they take the decision making power away from the addict. Chips take away one's free will. It enables a person to make a better choice not to take drugs at all.

X. FUTURE OF BRAIN CHIPS

Computerworld - By the year 2020, you won't need a keyboard and mouse to control your computer, say Intel Corp. researchers. Instead, users will open documents and surf the Web using nothing more than their brain waves.

Scientists at Intel's research lab in Pittsburgh are working to find ways to read and harness human brain waves so they can be used to operate computers, television sets and cell phones. The brain waves would be harnessed with Intel-developed sensors implanted in people's brains.

The scientists say the plan is not a scene from a sci-fi movie -- Big Brother won't be planting chips in your brain against your will. Researchers expect that consumers will want the freedom they will gain by using the implant.

Pomerleau said researchers are close to gaining the ability to build brain sensing technology into a head set that could be used to manipulate a computer. The next step is development of a tiny, far less cumbersome sensor that could be implanted inside the brain.Such brain research isn't limited to Intel and its university partners.

XI. CONCLUSION

The real application of brain chips lie in the medical world. It acts as a boon for the mentally and physically challenged people. Future could bring a revolution as brain computer interfaces are constructed using neurotechnology It offers the possibility unimaginable levels of independence for the severely disabled. This, in fact will be the perfect amalgamation of the machines and minds.

REFERENCES

[39] The micro chip that will save your memory:Scientists set to implant device to preserve experiences into BRAINS - by Daily mail Reporter Published:8May 2013
[40] http://members.tripod.com
[41] A New Chips Thinks Like a Brain- by Max Cacas Published:March 1,2013
[42] www.informationweek.com/story/IWK20020124S0026
[43] www.bu.edu/wcp/Papers/Bioe/BioeMcGe.htm
RF Based Inter Vehical Communication

Mr. Ambadas J. Gawande¹, Mr. Sunil W. Raut², Mr. Digambar S. Kale³

Department of Electronics and Telecommunication Engineering,
Dr. Bhauaheb Nandurkar College of Engineering & Technology, Yavatmal.

Abstract— Many accidents occur today when distant objects or roadway impediments are not quickly detected. To avoid these accidents, longer-range safety systems are needed with real time detection capability. In all the field tests, a single unidirectional radio link is evaluated, either between two vehicles or between a roadside unit and vehicle. To avoid road traffic collisions, vehicles will be required to periodically broadcast their position and speed to nearby vehicles. There are selected attributes to have the vehicular wireless safety system out of which the 902 MHz DSRC system is the only system that:

Is dedicated for transportation sector.(Licensed band)
1. Provides low latency using direct V2V links.
2. Provides broadband, real time, long range, and bi-directional communications.

In the above said method, to implement vehicle safety systems with one or two channels and using a single transceiver, consideration must be given to how a group of vehicles in a localized area becomes aware in real time of potentially dangerous situation which enables vehicles to use DSRC resources to synchronize with each other, receive safety messages with low transmission latency. Emerging systems for active vehicle safety use short range sensors with LOS links, usually to detect vehicles or lane boundaries immediately adjacent to the host vehicle. Typical applications include forward collision warning, adaptive cruise control, and lane keeping.

Keywords:
Radio frequency bands, MAC,ASDM, timeslot/code allocation, Bluetooth, Zigbee, IEEE 802.11p and P1609.

I. INTRODUCTION

Road and traffic safety can be improved if drivers have the ability to see further down the road and know if an collision has occurred, or if they are approaching a traffic jam. This can become possible if drivers and vehicles communicate with each other and with roadside base stations. If traffic information was provided to drivers, police, and other authorities, the roads would be safer and traveling on them would become more efficient. It is possible to build a multihop network among several vehicles that have communication devices. These vehicles would form a mobile ad hoc network, and could pass along information about road conditions, accidents, and congestion. A driver could be made aware of the emergency braking of a preceding vehicle or the presence of an obstacle in the roadway. Such a network could also help plotting vehicles (strings of vehicles that communicate with each other so they can maintain a tight inter-vehicle spacing) utilize thoroughways efficiently. It can also help vehicles negotiate critical points like blind crossings (intersections without traffic lights) and entries to highways.

II. LITERATURE REVIEW

Traffic accidents and highway congestion continues to remain a serious problem world-wide. Active safety applications, that use autonomous vehicle sensors such as radar, camera, etc., are being developed and deployed in vehicles by automakers to address the crash problem. Moreover, the FCC has recognized the importance of having a dedicated wireless spectrum for improving traffic safety and highway efficiency. In the US, the FCC has allocated 902 MHz band as Dedicated Short Range Communication (DSRC) for the primary purpose of improving transportation safety and highway efficiency. The reliability of DSRC vehicle-to-vehicle communication is adequate since packet drops do not occur in bursts most of the time. We also show that the application level reliability of VSC applications based on DSRC communication is quite satisfactory. Significant characteristics of DSRC communication for highly mobile vehicle-to-vehicle wireless network, which will contribute to better design and evaluation of communication protocols for VSC applications in future.

Emerging systems for active vehicle safety use short range sensors with LOS links, usually to detect vehicles or lane boundaries immediately adjacent to the host vehicle. Typical applications include forward collision warning, adaptive cruise control, and lane keeping. Longer-range vehicle safety systems are needed to help reduce accidents originating from more distant emergency events, roadway impediments, blind corners, and cross traffic. To detect these remote events, such systems may require up to 1000 meters of LOS coverage, and NLOS coverage to detect dangerous events ahead, but out of view.

In many cases, the ability to detect an emergency event occurring at some distance ahead is limited by the inability of drivers to see past the vehicle in front of them. The inability of drivers to react in time to emergency situations often creates a potential for chain collisions, in which an initial collision between two vehicles is followed by a series of collisions involving the following vehicles. The smaller frequency standard of DSRC i.e. 902 MHz can provide real-time alerts to drivers who cannot see remote or oncoming safety hazards.

The communication between two vehicles is governed by wireless protocol 802.11[4][7] Requirement of implementation of CCA is in the context of a vehicle-to-
vehicle wireless network, primarily at the Medium Access Control (MAC) and the routing layer.[2]

The 802.11 standard adopts the method of Carrier Sense Multiple Access with Collision Avoidance (CSMA/CA) and provides two access methods, Distributed Coordination Function (DCF) and Point Coordination Function (PCF).[5] DCF is a random access scheme based on carrier sense multiple accesses with collision avoidance (CSMA/CA). Before data transmission, it will detect the channel state in clearance or not. DCF is a required procedure of the 802.11 standard. In the DCF scheme, all stations content for the medium. If the medium is busy, each station runs a back-off algorithm to avoid collisions.[9]

PCF is an optional procedure of the 802.11 standard. In the PCF scheme, there is a central Point Coordinator (PC). The PC sends a beacon message to inform all stations to stop their DCF activities. As the PCF used, the medium will be partitioned into contention-free period (CFP) and contention period (CP) that CFP is coordinated by PCF and CP is DCF. The PC then polls each station for data transmission. During this Contention Free Period (CFP), stations are not allowed for data transmission until they are polled.[6]

### III. METHODOLOGY USED IN THIS PAPER

In this paper we are using three terminologies, which are DSRC, MAC Protocol, and Micro-controller 16f877A.

1) DSRC (Dedicated Short Range Communication)

This paper discusses the technical requirements of a new wireless standard, Dedicated Short Range Communications (DSRC), and its applications in supporting ITS. In recognition of this need, Federal Communication Commission (FCC) allocated 12 MHz spectrum in the 902MHz band for DSRC in 1998. The DSRC standard supports vehicles with an on-board device (OBD) to communicate with a roadside unit (RSU), or other traveling vehicles. FCC provides several examples of DSRC applications: “travellers” alerts, automatic toll collection, traffic congestion detection, emergency dispatch services, and electronic inspection of moving trucks through data transmissions with roadside inspection facilities.” We classify these applications into unicast (one sender and one receiver) vs. broadcast (one sender and many receivers) and RSU-to-Vehicle (R2V) vs. Vehicle-to-Vehicle (V2V).

<table>
<thead>
<tr>
<th>Items</th>
<th>Legacy</th>
<th>New</th>
</tr>
</thead>
<tbody>
<tr>
<td>Band (MHz)</td>
<td>900-928</td>
<td>5.850-5.925 GHz</td>
</tr>
<tr>
<td>Spectrum (MHz)</td>
<td>12</td>
<td>75</td>
</tr>
</tbody>
</table>

The DSRC standard supports vehicles with an on-board device (OBD) to communicate with a roadside unit (RSU), or other traveling vehicles.

### IV. GENERALIZED BLOCK DIAGRAM VEHICLES

![Block Diagram Vehicles](image)

### V. OBSTACLE DETECTOR

This unit consists of IR sensors which mean the IR transmitter and IR receiver. Astable multivibrator is used at the input side of the IR transmitter. This astable multi vibrator will make the IR ray to emit continuously.

IC555 timer is used as an astable multivibrator. IR receiver is provided with the 5 volt battery. When the IR is intermittent then an obstacle is detected.

Switches:

Switches are used to provide trigger to micro-controller AT MEGA 16. There are six different alert messages which are stored at certain memory location in micro-controller and there are six switches for each of them. Whenever we operate the switch manually, the corresponding message is displayed and transmitted to the receiver through transmitter. A push-to-make switch is used for this operation. A push-to-make switch returns to its normally open (off) position when you release the button.

### VI. AMPLIFIERS

There are two amplifier circuits are used in the block diagram

1) Amplifier circuit before RF transmitter:

We have to supply 9V signal to RF transmitter and the input signal to the amplifier is either 5Volts or 0Volts as the signals coming from the IC AT MEGA 16 is
digital in which 1 is equivalent to 5V & 0 is equivalent to 0V. So we have to use amplifier circuit before RF transmitter.

2) Amplifier circuit after RF transmitter:
Strength of the received signal which is transmitted from the RF transmitter has been reduced. Therefore it is necessary to regain it by means of amplification.RF Transmitter & Receiver. Radio Frequency transmitter and receiver are being used for proper transmission & reception of the signals. A 9Volts supply is given to this section. RF transmitter and receiver module provides frequency of 433 MHz

VII. LCD DISPLAY

The two LCD displays which are being used in two vehicles are 16 Character x 2 Line LCDs. These are used to display different alert messages. The most common type of LCD controller is the Hitachi 44780 which provides a relatively simple interface between a microcontroller and an LCD.

VIII. CIRCUIT EXPLANATION

1) Power Supply:
The simple 5V power supply used in the circuits is having following features:

- Brief description of operation: Gives out well regulated +5V output, output current capability of 100 mA
- Circuit protection: Built-in overheating protection shuts down output when regulator IC gets too hot
- Circuit complexity: Very simple and easy to build
- Circuit performance: Very stable +5V output voltage, reliable operation
- Availability of components: Easy to get, uses only very common basic components
- Design testing: Based on datasheet example circuit, I have used this circuit successfully as part of many electronics papers
- Applications: Part of electronics devices, small laboratory power supply
- Power supply voltage: Unregulated DC 8-18V power supply
- Power supply current: Needed output current +5 V.

2) Circuit Description:

This circuit is a small +5V power supply, which is useful when experimenting with digital electronics. Small inexpensive wall transformers with variable output voltage are available from any electronics shop and supermarket. Those transformers are easily available, but usually their voltage regulation is very poor, which makes then not very usable for digital circuit experimenter unless a better regulation can be achieved in some way. This circuit can give +5V output at about 150 mA current, but it can be increased to 1 A when good cooling is added to 7805 regulator chip. The circuit has over load and terminal protection. The capacitors must have enough high voltage rating to safely handle the input voltage feed to circuit. Pin out of the 7805 regulator IC.

1. Unregulated voltage in
2. Ground
3. Regulated voltage out

3) Component list:
- 7805 regulator IC
- 100 uF electrolytic capacitor, at least 25V voltage rating.
- 10 uF electrolytic capacitor, at least 6V voltage rating.
- 100 nF ceramic or polyester capacitor.

4) Other output voltages:
If we need other voltages than +5V, you can modify the circuit by replacing the 7805 chips with another regulator with different output voltage from regulator 78xx chip family. The last numbers in the chip code tells the output voltage. Remember that the input voltage must be at least 3V greater than regulator output voltage otherwise the regulator does not work well.

5) Features:
- Output Current up to 1A.
- Output Voltages of 5, 6, 8, 9, 10, 12, 15, 18, 24V.
- Thermal Overload Protection.
- Short Circuit Protection.
- Output Transistor Safe Operating Area Protection.

6) Description:
The KA78XX/KA78XXA series of three-terminal positive regulator are available in the TO-220/D-PAK package and with several fixed output voltages, making them useful in a wide range of applications. Each type employs internal current limiting, thermal shut down and safe operating area protection, making it essentially indestructible. If adequate heat sinking is provided, they can deliver over 1A output current. Although designed primarily as fixed voltage regulators, these devices can be
used with external components to obtain adjustable voltages and currents.

IX. INFRA-RED SENSORS

It includes IR LEDs, an Op amp, a transistor and a couple of resistors. In need, as the title says, a standard IR led is used for the purpose of detection. Due to that fact, the circuit is extremely simple, and any novice electronics hobbyist can easily understand and build it.

It is the same principle in ALL Infra-Red proximity sensors. The basic idea is to send infra red light through IR-LEDs, which is then reflected by any object in front of the sensor. Then all you have to do is to pick-up the reflected IR light. For detecting the reflected IR light, we are going to use a very original technique: we are going to use another IR-LED, to detect the IR light that was emitted from another led of the exact same type This is an electrical property of Light Emitting Diodes (LEDs) which is the fact that a led Produce a voltage difference across its leads when it is subjected to light. As if it was a photo-cell, but with much lower output current. In other words, the voltage generated by the led can't be - in any way - used to generate electrical power from light, It can barely be detected. that's why as you will notice in the schematic, we are going to use a Op-Amp (operational Amplifier) to accurately detect very small voltage changes.

X. RELAYS

Relays are electromagnetically operated, remote controlled switches, with one or more sets of contacts. When energized, the relay operates to open or close its contacts or to open contacts and close other .Contacts, which are open when the relay is not energized, are called “Normally open or simply open contacts”, contacts which are closed when the relay is not energized, are called “Normally closed contacts”.

The basis for relays, is the simplest relay, is the single pole, single throw (SPST) relay .it is nothing more than an electromagnetically controlled on-off switch. This is desirable because we can now use smaller diameter wires, to control the current flow through a much larger wire, and also to limit the wear and tear of control switch. The basis for relays is the simple electromagnet.The simplest relay is the Single Pole, Single Throw (spst) relay. It is nothing more than an electromagnetically controlled, to control the current flow through a much larger wire, and also to limit the wear and tear on the control switch.

XI. DC Motors

1) Basics of D.C Motors:

A direct current (DC) motor is a fairly simple electric motor that uses electricity and a magnetic field to produce torque, which turns the motor. At its most simple, a DC motor requires two magnets of opposite polarity and an electric coil, which acts as an electromagnet. The repellent and attractive electromagnetic forces of the magnets provide the torque that causes the DC motor to turn. A DC motor requires at least one electromagnet. This electromagnet switches the current flow as the motor turns, changing its polarity to keep the motor running. The other magnet or magnets can either be permanent magnets or other electromagnets. Often, the electromagnet is located in the center of the motor and turns within the permanent magnets, but this arrangement is not necessary.

2) Principle of Operation:

The principle of any electric motor is based upon Ampere’s law which states that “when a current carrying conductor is placed in a magnetic field, it experiences a
mechanical force.” This magnetic field may be created by “field coil” wound on stator by permanent magnet.

The direction is defined by the “Right Hand Rule”. Place the thumb along the wire, in the direction of the following current. When we wrap our fingers around the wire, the direction of the magnetic field is in the direction of our fingers. When a wire & its magnetic field are placed into another uniform magnetic field, the magnetic field interacts in a way that puts a force on the wire. The amount of force is defined by the formula:

\[ F = B \cdot I \cdot L \]

Where,  
- \( F \) = force (N-Newton)  
- \( B \) = magnetic field strength (T-tesla)  
- \( I \) = current (A-ampere)  
- \( L \) = Length of wire in magnetic field (meters)

This force is basis behind the driving force inside a DC motor. But this will lead to infinite velocity. To overcome this problem there is another factor given by Faraday Law.

“When you have a motor moving in a magnetic field, an EMF (electromotive force voltage) is created in the motor. This is called faradays law, & the EMF is proportional to the rate of change of the magnetic field.”

\[ V_{emf} = N \frac{d\Phi}{dt} \]

Where, \( V_{emf} \) = voltage against current in wire (v-volts)  
\( N \) = number of wires  
\( \frac{d\Phi}{dt} \) = time rate of change of magnetic flux.

This back EMF is opposite to the voltage that is driving the current through the motor. As the motor speeds up, the magnetic flux it encounters starts to change, & the back EMF begins to slow the amount of current flowing through the motor. This process keeps going until the back EMF almost cancels the voltage supplied across the motor. This is the points where the points where the motor reaches a common velocity. When a load to the motor, this slows the motor a little, & decreases the flux, increasing the force to balance the torque.

Above is the quite simple schematic of LCD. The LCD panel's Enable and Register Select is connected to the Control Port. The Control Port is an open collector open drain output. While most Parallel Ports have internal pull-up resistors, there is a few which don't. Therefore by incorporating the two 10K external pull up resistors, the circuit is more portable for a wider range of computers, some of which may have no internal pull up resistors. We make no effort to place the Data bus into reverse direction. Therefore we hard wire the R/W line of the LCD panel, into write mode. This will cause no bus conflicts on the data lines. As a result we cannot read back the LCD's internal Busy Flag which tells us if the LCD has accepted and finished processing the last instruction. This problem is overcome by inserting known delays into our program. The 10k Potentiometer controls the contrast of the LCD panel. Nothing fancy here. As with all the examples, I've left the power supply out. You can use a bench power supply set to 5v or use a on board +5 regulator.

<table>
<thead>
<tr>
<th>TABLE II PIN ASSIGNMENT</th>
</tr>
</thead>
<tbody>
<tr>
<td>Pins</td>
</tr>
<tr>
<td>1</td>
</tr>
<tr>
<td>2</td>
</tr>
<tr>
<td>3</td>
</tr>
<tr>
<td>4</td>
</tr>
<tr>
<td>5</td>
</tr>
<tr>
<td>6</td>
</tr>
<tr>
<td>7 – 14</td>
</tr>
</tbody>
</table>
XIII. CONCLUSIONS

In this above discussion we are conclude that the system based on RF based inter vehicles communication has wide application such as 
- Reduces traffic congestion,
- Reduces accident rate,
- Reduces injuries and fatalities

Provides direct vehicle to vehicle communication ,Real time communication .In future we can implement this system on GSM as well as GPS.

REFERENCES


“Topology Control in Mobile Ad-Hoc Network with Co-Operative Communication”

Amol K. Tholbare¹, Raul D. Maske², Prof. Sandip Asutkar³
¹(Final year B.E (Information Technology), DBNCOET Yavatmal, India)
²(Third Year B.E (Mechanical), DBNCOET Yavatmal, India)
³(Asst. Prof., DBNCOET Yavatmal, India)

ABSTRACT
Cooperative-communication has received tremendous interests in wireless network most existing works on cooperative communications are focused on link-level physical layer issues. Consequently, the impacts of cooperative communications on network-level upper layer issues, such as topology control, routing and network capacity, are largely ignored. In this paper, we propose a Capacity-Optimized Cooperatives (COCO) topology control scheme to improve the network capacity in MANETs by jointly considering both upper layer network capacity and physical layer cooperative communications. Using simulation examples, we show that physical layer cooperative communications have significant impacts on the performance of topology control and network capacity, and the proposed topology control scheme can substantially improve the network capacity in MANETs with cooperative communications.

Keywords: MANET, AD-HOC, COOPERATIVE COMMUNICATION.

1. Introduction

1.1 Mobile Ad Hoc Networks (MANETs)
Mobile ad-hoc network is an independent system of mobile nodes connected by wireless links forming a short, live, on-the-fly network even when access to the Internet is unavailable. Nodes in MANETs generally operate on low power battery devices. These nodes can function both as hosts and as routers. As a host, nodes function as a source and destination in the network and as a router, nodes act as intermediate bridges between the source and the destination giving store-and-forward services to all the nodes.

1.2 The topology control problem in MANET
In mobile ad hoc wireless communication, each node of the network has a potential of varying the topology through the adjustment of its power transmission in relation to other nodes in the neighborhood. In contrast, wired networks have fixed established pre-configured infrastructure with centralized network management system structure in place. Therefore, the fundamental reason for the topology control scheme in MANET is to provide a control mechanism that maintains the network connectivity and performance optimization by prolonging network lifetime and maximizing network throughput. A MANET topology can depend on uncontrollable factors such as node mobility, weather, interference, noise as well as controllable factors such as transmission power, directional antennas and multi-channel communications.
A bad topology can impact negatively on the network capacity by limiting spatial reuse capability of the communication channel and also can greatly undermine the robustness of the network. Where network capacity means the bandwidth and ability for it to be used for communication. A network partitioning can occur in a situation where the network topology becomes too sparse. Similarly, a network which is too dense is prone to interference at the medium access (MAC) layer, the physical layer of the network. So the network should neither be too dense nor too sparse for efficient communication amongst nodes to take place.

2. Existing System

Most existing works are focused on link-level physical layer issues, such as outage probability and outage capacity. Consequently, the impacts of cooperative communications on network-level upper layer issues, such as topology control, routing and network capacity, are largely ignored. Indeed, most of current works on wireless networks attempt to create, adapt, and manage a network on a maze of point-to-point non-cooperative wireless links. Such architectures can be seen as complex networks of simple links.

3. Proposed System

We propose a Capacity-Optimized Cooperative (COCO) topology control scheme to improve the network capacity in MANETs by jointly considering both upper layer network capacity and physical layer cooperative communications. Through simulations, we show that physical layer cooperative communications have significant impacts on the network capacity, and the proposed topology control scheme can substantially improve the network capacity in MANETs with cooperative communications.

4. Network Constraints

With physical layer cooperative communications, direct transmissions, multi-hop transmissions and cooperative transmissions. Direct transmissions and multi-hop transmissions can be regarded as special types of cooperative transmissions. A direct transmission utilizes no relays while a multi-hop transmission does not combine signals at the destination. In Fig.1, the cooperative channel is a virtual multiple-input single-output (MISO) channel, where spatially distributed nodes are coordinated to form a virtual antenna to emulate multi antenna transceivers. Two constraint conditions need to be taken into consideration in the proposed COCO topology control scheme.

5. Related Work

5.1 COOPERATIVE COMMUNICATIONS

Cooperative communication typically refers to a system where users share and coordinate their resources to enhance the information transmission quality. It is a generalization of the relay communication, in which multiple sources also serve as relays for each other. Early study of relaying problems appears in the information theory community to enhance communication between the source and destination. Recent tremendous interests in cooperative communications are due to the increased understanding of the benefits of multiple antenna systems. Although multiple-input multiple-output (MIMO) systems have been widely acknowledged, it is difficult for some wireless mobile devices to support multiple antennas due to the size and cost constraints. Recent studies show that cooperative communications allow single antenna devices to work together to exploit the spatial diversity and reap the benefits of MIMO systems such as resistance to fading, high throughput, low...
transmitted power, and resilient networks. In a simple cooperative wireless network model with two hops, there is a source, a destination, and several relay nodes. The basic idea of cooperative relaying is that some nodes, which overheard the information transmitted from the source node, relay it to the destination node instead of treating it as interference. Since the destination node receives multiple independently faded copies of the transmitted information from the source node and relay nodes, cooperative diversity is achieved. Relaying could be implemented using two common strategies.

- **Amplify-and-forward**
- **Decode-and-forward**

In amplify-and-forward, the relay nodes simply boost the energy of the signal received from the sender and retransmit it to the receiver. In decode-and-forward, the relay nodes will perform physical-layer decoding and then forward the decoding result to the destinations. If multiple nodes are available for cooperation, their antennas can employ a space-time code in transmitting the relay signals. It is shown that cooperation at the physical layer can achieve full levels of diversity similar to a MIMO system, and hence can reduce the interference and increase the connectivity of wireless networks. Most existing works about cooperative communications are focused on physical layer issues, such as decreasing outage probability and increasing outage capacity, which are only link-wide metrics. However, from the network’s point of view, it may not be sufficient for the overall network performance, such as the whole network capacity. Therefore, many upper layer network-wise metrics should be carefully studied, e.g., the impacts on network structure and topology control. Cooperation offers a number of advantages in flexibility over traditional wireless networks that go beyond simply providing a more reliable physical layer link. Since cooperation is essentially a network solution, the traditional link abstraction used for networking design may not be valid or appropriate. From the perspective of a network, cooperation can benefit not only the physical layer, but the whole network in many different aspects. With physical layer cooperative communications, there are three transmission manners in MANETs: direct transmissions (Fig.1), multi-hop transmissions and cooperative transmissions (Fig.1). Direct transmissions and multi-hop transmissions can be regarded as special types of cooperative transmissions. A direct transmission utilizes no relays while a multi-hop transmission does not combine signals at the destination. In Fig. 1c, the cooperative channel is a virtual multiple-input single-output (MISO) channel, where spatially distributed nodes are coordinated to form a virtual antenna to emulate multi-antenna transceivers.

### 5.2 TOPOLOGY CONTROL

The destination combines the two signals from the source and the relay to decode the information. The network topology in a MANET is changing dynamically due to user mobility, traffic, node batteries, and so on. Meanwhile, the topology in a MANET is controllable by adjusting some parameters such as the transmission power, channel assignment, etc. In general, topology control is such a scheme to determine where to deploy the links and how the links work in wireless networks to form a good network topology, which will optimize the energy consumption, the capacity of the network, or end-to-end routing performance. Topology control is originally developed for wireless sensor networks (WSNs), MANETs, and wireless mesh networks to reduce energy consumption and interference. It usually results in a simpler network topology with small node degree and short transmission radius, which will have high-quality links and less contention in medium access control (MAC) layer. Spatial/spectrum reuse will become possible due to the smaller radio coverage. Other properties like symmetry and planarity are expected to obtain in the resultant topology. Symmetry can facilitate wireless communication and two-way handshake schemes for link acknowledgment while planarity increases the possibility for parallel transmissions and space reuse. Power control and channel control issues are coupled with topology control in MANETs while they are treated separately traditionally. Although a mobile
node can sense the available channel, it lacks of the scope to make network wide decisions. It therefore makes more sense to conduct power control and channel control via the topological viewpoint. The goal of topology control is then to set up interference-free connections to minimize the maximum transmission power and the number of required channels. It is also desirable to construct a reliable network topology since it will result in some benefits for the network performance. Topology control focuses on network connectivity with the link information provided by MAC and physical layers. There are two aspects in a network topology: network nodes and the connection links among them. In general, a MANET can be mapped into a graph G(V, E), where V is the set of nodes in the network and E is the edge set representing the wireless links. A link is generally composed of two nodes which are in the transmission range of each other in classical MANETs. The topology of such a classical MANET is parameterized by some controllable parameters, which determine the existence of wireless links directly. In traditional MANETs without cooperative communications, these parameters can be transmit power, antenna directions, etc. In MANETs with cooperative communications, topology control also needs to determine the transmission manner (i.e., direct transmission, multi-hop transmission, or cooperative transmission) and the relay node if cooperative transmission is in use. As topology control is to determine the existence of wireless links subject to network connectivity, the general topology control problem can be expressed as

\[ G^* = \arg \max f(G) \]  

(1)

The problem Eq. 1 uses the original network topology G, which contains mobile nodes and link connections, as the input. According to the objective function, a better topology G*(V, E*) will be constructed as the output of the algorithm. G* should contain all mobile nodes in G, and the link connections E* should preserve network connectivity without partitioning the network. The structure of resulting topology is strongly related to the optimization objective function, which is f(G) in Eq. 1. It is difficult to collect the entire network information in MANETs. Therefore, it is desirable to design a distributed algorithm, which generally requires only local knowledge, and the algorithm is run at every node independently. Consequently, each node in the network is responsible for managing the links to all its neighbours’ only. If all the neighbour connections are preserved, the end-to-end connectivity is then guaranteed. Given a neighbourhood graph GN (VN, EN) with N neighbouring nodes, we can define a distributed topology control problem as \[ G^*N = \arg \max f(GN) \], s.t. connectivity to all the neighbours’. The objective function f(G) in Eq. 1 is critical to topology control problems. Network capacity is an important objective function. Our previous work [8] shows that topology control can affect network capacity significantly. In the following section, we present a topology control scheme with the objective of optimizing network capacity in MANETs with cooperative communications.

6. TOPOLOGY CONTROL FOR NETWORK CAPACITY IMPROVEMENT IN MANETS WITH COOPERATIVE COMMUNICATIONS

In this section, we first describe the capacity of MANETs. Then, we present the proposed COCO topology control scheme for MANETs with cooperative communications. THE CAPACITY OF MANETS As a key indicator for the information delivery ability, network capacity has attracted tremendous interests since the landmark paper. There are different definitions for network capacity. Two types of network capacity are introduced in the first one is transport capacity, which is similar to the total one-hop capacity in the network. It takes distance into consideration and is based on the sum of bit-meter products. One bit-meter means that one bit has been transported to a distance of one meter toward its destination. Another type of capacity is throughput capacity, which is based on the information capacity of a channel. Obviously, it is the amount of all the data successfully transmitted during a unit time. It has been shown that the capacity in wireless ad hoc networks is limited. In traditional MANETs without cooperative communications, the capacity is decreased as the number of nodes in the network increases. Asymptotically, the per-node throughput declines to zero when the number of nodes approaches to infinity. In this study, we adopt the second type of definition. The expected network capacity is determined by various factors: wireless channel data rate in the physical layer, spatial reuse scheduling and interference in the link layer, topology control presented earlier, traffic balance in routing, traffic patterns, etc. In the physical layer, channel data rate is one of the main factors. Theoretically, channel capacity can be derived using Shannon’s capacity formula. In practice, wireless channel data rate is jointly determined by the modulation, channel coding, transmission power, fading, etc. In addition, outage capacity is usually used in practice, which is supported by a small outage probability, to represent the link capacity. In the link layer, the spatial reuse is the major ingredient that affects network capacity. Link interference, which refers to the affected nodes during the
transmission, also has a significant impact on network capacity. Higher interference may reduce simultaneous transmissions in the network, thus reduce the network capacity, and vice versa. The MAC function should avoid collision with existing transmission. It uses a spatial land temporal scheduling so that simultaneous transmissions do not interfere with each other. Nodes within the transmission range of the sender must keep silent to avoid destroying ongoing transmissions. In addition, there are some factors that prevent the channel capacity from being fully utilized, such as hidden and exposed terminals, which need to be solved using handshake protocols or a dedicated control channel in wireless networks. Routing not only finds paths to meet quality of service (QoS) requirements, but also balances traffic loads in nodes to avoid hot spots in the network. By balancing traffic, the network may admit more traffic flows and maximize the capacity. Since we focus on topology control and cooperative communications, we assume an ideal load balance in the network, where the traffic loads in the network are uniformly distributed to the nodes in the network. The study shows that cooperative transmissions do not always outperform direct transmissions. If there is no such relay that makes cooperative transmissions have larger outage capacity, we rather transmit information directly or via multi-hops. For this reason, we need to determine the best link block and the best relay to optimize link capacity. On the other hand, other nodes in the transmission range have to be silent in order not to disrupt the transmission due to the open shared wireless media. The affected areas include the coverage of the source, the coverage of the destination, as well as the coverage of the relay.

7. IMPROVING NETWORK CAPACITY USING TOPOLOGY CONTROL IN MANETS WITH COOPERATIVE COMMUNICATIONS

To improve the network capacity in MANETs with cooperative communications using topology control, we can set the network capacity as the objective function in the topology control problem in Eq. 1. In order to derive the network capacity in a MANET with cooperative communications, we need to obtain the link capacity and inference model when a specific transmission manner (i.e., direct transmission, multi-hop transmission, or cooperative transmission) is used. When traditional direct transmission is used, given a small outage probability, the outage link capacity can be derived. Since only two nodes are involved in the direct transmission, the interference set of a direct transmission is the union of coverage sets of the source node and the destination node. In this article, we adopt the interference model in which confines concurrent transmissions in the vicinity of the transmitter and receiver. This model fits the medium access control function well (e.g., the popular IEEE 802.11 MAC in most mobile devices in MANETs). Herein, interference of a link is defined as some combination of coverage of nodes involved in the transmission. Multi hop transmission can be illustrated using two-hop transmission. When two-hop transmission is used, two time slots are consumed. In the first slot, messages are transmitted from the source to the relay, and the messages will be forwarded to the destination in the second slot. The outage capacity of this two-hop transmission can be derived considering the outage of each hop transmission. The transmission of each hop has its own interference, which happens in different slots. Since the transmissions of the two hops cannot occur simultaneously but in two separate time slots, the end-to-end interference set of the multi-hop link is determined by the maximum of the two interference sets. When cooperative transmission is used, a best relay needs to be selected proactively before transmission. In this study, we adopt the decode and forward relaying scheme. The source broadcasts its messages to the relay and destination in the first slot. The relay node decodes and recodes the signal from the source, and then forwards it to the destination in the second slot. The two signals of the source and the relay are decoded by maximal rate combining at the destination. The maximum instantaneous end-to-end mutual information, outage probability, and outage capacity can be derived. For the interference model, in the broadcast period, both the covered neighbours’ of the source and the covered neighbours of the relay and the destination have to be silent to ensure successful receptions.

FIG3: The original topology: a MANET with 30 nodes randomly deployed in a 800 x 800 m² area.
FIG5. The final topology generated by COCO.

The solid lines denote traditional direct transmissions and multi hop transmissions. The dashed lines denote the links involved in cooperative communications. In the second slot, both the covered neighbours of the selected relay and the destination have to be silent to ensure successful receptions. After obtaining the link capacity and inference models, the network capacity can be derived [8] as the objective function in the topology control problem in Eq.1. By considering direct transmission, multi hop transmission, cooperative transmission, and interference, the proposed COCO topology control scheme extends physical layer cooperative communications from the link-level perspective to the network-level perspective in MANETs. The proposed scheme can determine the best type of transmission and the best relay to optimize network capacity. Two constraint conditions need to be taken into consideration in the proposed COCO topology control scheme. One is network connectivity, which is the basic requirement in topology control. The end-to-end network connectivity is guaranteed via a hop-by-hop manner in the objective function. Every node is in charge of the connections to all its neighbour’s. If all the neighbour connections are guaranteed, the end-to-end connectivity in the whole network can be preserved. The other aspect that determines network capacity is the path length. An end-to-end transmission that traverses more hops will import more data packets into the network. Although path length is mainly determined by routing, COCO limits dividing a long link into too many hops locally. The limitation is two hops due to the fact that only two-hop relaying is adopted.

8. RESULTS AND DISCUSSIONS

In this section, the performance of the proposed scheme is illustrated using computer simulations. We consider a MANET with 30 nodes randomly deployed in a 800 × 800 m² area. The number of nodes is changed in the simulations. The channels follow a Raleigh distribution. We compare the performance of the proposed scheme with that of an existing well-known topology control scheme, called LLISE, which only considers traditional multi-hop transmissions without cooperative communications and preserves the minimum interference path for each neighbour link locally. We also show the worst network capacity among all the topology configurations for comparison. The original topology is shown in Fig. 2, where links exist whenever the associated two end nodes are within transmission range of each other. It is clear that this topology lacks any physical layer cooperative communications. Figure 3 shows the resulting topology using the proposed COCO topology control scheme. In Fig. 3, the solid lines denote traditional direct transmissions and multi-hop transmissions, and the dash lines denote links involved in cooperative communications. As we can see from Fig. 3, to maximize the network capacity of the MANET, many links in the network are involved in cooperative communications. One example of two-phase cooperative communications is shown in the top left corner of the figure. Figure 4 shows the network capacity per node in different topology control schemes with different numbers of nodes in the MANET.

Figure 4. Network capacity versus different numbers of nodes in the MANET.

As we can see from the figure, the proposed COCO scheme has the highest network capacity regardless of the number of nodes in the network. Similar to COCO, LLISE is executed in each node distributed. It preserves all the edges on the minimum interference path for each link in the resulting topology, thus minimizes the interference to improve network capacity. Nevertheless, COCO can achieve a much higher network capacity than LLISE, since LLISE only considers multihop transmissions. The performance gain of the proposed scheme comes from the joint design of transmission mode selection, relay node selection, and interference minimization in MANETs with cooperative communications.
9. CONCLUSIONS AND FUTURE WORK
In this article, we have introduced physical layer cooperative communications, topology control, and network capacity in MANETs. To improve the network capacity of MANETs with cooperative communications, we have proposed a Capacity-Optimized Cooperative (COCO) topology control scheme that considers both upper layer network capacity and physical layer relay selection in cooperative communications. Simulation results have shown that physical layer cooperative communications techniques have significant impacts on the network capacity, and the proposed topology control scheme can substantially improve the network capacity in MANETs with cooperative communications. Future work is in progress to consider dynamic traffic patterns in the proposed scheme to further improve the performance of MANETs with cooperative communications.

10. REFERENCES
[1]. Quansheng Guan, South China University of Technology F. Richard Yu, Carleton University, Ottawa Shengming jiang, South China University of Technology victor c. Leung, university of British Columbia Hamid Mehrvar, Ciena Inc.”TOPOLOGY CONTROL IN MOBILE AD HOC NETWORKS WITH COOPERATIVE COMMUNICATIONS”.


[3]. T.S.Asha Associate Professor Department of ECEENSS College of Engineering, Palakkad, Kerala,” NETWORK CONNECTIVITY BASED TOPOLOGY CONTROL FOR MOBILE AD HOC NETWORKS”


