Abstract—Mobile ad hoc networks (MANET) consists of group of mobile nodes which may communicate without any form of pre existing infrastructures, these nodes exhibit dynamic topology and random movement which causes variations in link capacity. Routing traffic between MANET nodes experience collisions, end to end delay (E2E) and total loss of data packets due to limited bandwidth between nodes. These result in poor Quality of Service (QoS) and E2E delay throughput. In this paper we have proposed the application of time division multiple access (TDMA) technique to improve data rate and reduce control and data overhead collisions. We compared the existing normal routing scheme and the TDMA based scheme. TDMA allows a number of clients to access a single radio-frequency channel without interference by allocating unique time slots to each user within each channel, reducing the loss of packets and improving the data rate thereby delivering QoS to the clients.

Index Terms— Energy Conservation, Routing, Time Division multi-access.

I. INTRODUCTION
Routing traffic between MANET nodes experience collisions, E2E and total loss of data packets due to limited bandwidth between nodes. These result in poor QoS and delay throughput. The transmission time is delayed when collision occurs resulting in an increase in the contention window within a multi hop transmission channel leading to a decrease in data rates. However, In this paper we investigated how to improve the data rate and reduce the effect of these collisions for a reliable end to end throughput by proposing a time division multiple access (TDMA) technique based on Binary Exponential Back off Algorithm (BEBA). Our analysis showed that this scheme when compared with the existing normal route technique provides a considerable improvement in control and over head data rates, as well as reduces the loss of data packet using the available bandwidth.

II. REVIEW OF THE STATE OF ART
In Mobile ad hoc networks, the key performance issues also referred to as MANET performance issues are based on the quantitative metrics that can be used to assess the performance of any routing protocol or traffic. These issues include E2E data throughput and delay, route acquisition time, control and data packets as well as protocol efficiency etc. However the critical issue or problem in this paper is that of low data throughput and delay due to heavy loss rates [1] and effect of collision and high retransmission rate.

Due to E2E reliability requirements in many MANET applications, owing to variations in network size, network connectivity, topological changes and link capacity, it has become so difficult to improve the Data rate, reduce the total loss of control packets and maintain better QoS, through normal routing using a limited single channel capacity. However in the light of the above, we wish to provide solutions to the research question;

- How can we improve the data rate and reduce collision in mobile ad hoc networks other than direct traffic generation within the available bandwidth?

In this paper our main contribution focuses on TDMA technique to improve the data rate and reduce the effect of collision in the network. This scheme has the following characteristic as main contributions;

- Substantial improvement in control and over head data rates.
data rates in the network.

- Substantial improvement in the network capacity utilization through multi-hop transmission.
- The TDMA technique is computationally efficient subject to time division aggregation and transmission of packets.
- Effective improvement in Bandwidth utilization and collision reduction.

IV. PROBLEM SOLUTION

Our TDMA model is based on a simulation approach using the Network Simulator-2 (NS-2) which is a discrete event simulator. NS-2 provides substantial support for simulation of routing, protocols over MANET. We investigated a performance evaluation to compare the loss rate and packet dropped.

Binary Exponential Backoff Algorithm (BEBA) is implemented (code available) which adjusts the contention window size dynamically in reaction to collision intensity. Such an algorithm is embedded in the IEEE 802.11 Distributed Coordination Function (DCF) [4]. We then applied TDMA technique by allocating a unique time slot for each station; thereby no station transmission is interfered. Advance On-demand Distance Vector (AODV) routing protocol was used based on dynamic topology of the nodes. In this model, 16-node logical link structure consisting of 2 scenarios was configured using the NS-2. The normal scenario which we called normal-16 BEBA is compared with the TDMA-16 BEBA. We generated the traffic of 50 bytes based on constant bit rate generator within the application layer, and a time interval of 0.01s between packets transmission was set which gives data rate of 40kbits/s. We observed the performance of the control packets in terms of data throughput and loss rates as shown in fig.1. The performance showed a considerable decrease in loss rates using our TDMA technique. Table 1. Shows the comparative numerical values of the two schemes.

Table 1. Difference in values of Data received and loss rates

<table>
<thead>
<tr>
<th>Nodes</th>
<th>NORMAL_16_BEBA</th>
<th>TDMA_16_BEBA</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>Data Received (Mbits/s)</td>
<td>Loss Rate (Mbits/s)</td>
</tr>
<tr>
<td>0-1</td>
<td>22.3</td>
<td>1.4</td>
</tr>
<tr>
<td>2-3</td>
<td>30.3</td>
<td>0.3</td>
</tr>
<tr>
<td>4-5</td>
<td>33.4</td>
<td>0.3</td>
</tr>
<tr>
<td>6-7</td>
<td>33.5</td>
<td>0.3</td>
</tr>
<tr>
<td>8-9</td>
<td>34.0</td>
<td>0.3</td>
</tr>
<tr>
<td>10-11</td>
<td>28.7</td>
<td>0.5</td>
</tr>
<tr>
<td>12-13</td>
<td>32.6</td>
<td>0.3</td>
</tr>
<tr>
<td>14-15</td>
<td>29.0</td>
<td>0.3</td>
</tr>
</tbody>
</table>

A comparison of data received (Odd number of nodes) and the corresponding loss rates (Even number of nodes) obtained with the Normal and TDMA technique.

V. CONCLUSION

We have proposed TDMA, a BEBA based time dependent data rate improvement technique in MANET, which characterizes improvement of data by reducing loss of control packet and collision effect over a limited channel capacity. We observe inefficient data throughput and E2E delay within a multi hop transmission between mobile nodes as an inherent problem.

This scheme, governed by packets generation and transmission over allocated time slot using the available bandwidth guarantees appreciable reduction in data loss rate and substantial improvement in link capacity utilization. The numerical results obtained shows that the performance of TDMA technique is better when compared to the current Normal route technique.

However, these results are subject to variations based on simulation environment, system clock and speed and may not reflect a standard practical representation of the technical analysis. In a related future work we propose an ideal simulation environment incorporating BEBA based on practical test bed on real time analysis. Future modifications to this technique are welcome.

Acknowledgment

Special thanks to our Supervisor Prof. Wlodek Kulesza, for his great ideas and support, and to our families and friends for their encouragement.

References


The Car Security System Switching Over GSM and RF
Muhammad A. R. Choudhary, Venkata R. V. S. Veguru, and Lokeshwara R. Bandaru

Abstract - There were three main problems in the car security systems which we are using now a days, these are range, backup and cost issues. After some analysis of different car security systems, we proposed to make a combination of Global System for Mobile Communications (GSM) and Radio Frequency (RF) Systems to overcome the above mentioned issues.

Index Terms - CAR SECURITY SYSTEMS, GSM SECURITY SYSTEM, RF SECURITY SYSTEM, and TRACKING SYSTEMS

I. INTRODUCTION

Car theft is most often crime of opportunity in present days. It has been increased due to lack of proper security measures. On average of every 23 seconds a break-in and theft happens. It is therefore necessary to minimize this risk. One way to reduce the probability of your vehicle being stolen is to install it with a quality alarm system. An alarm system combined with mobile technology improves the security of the car. It also tracks the position of the car and ensures its recovery if stolen.

Wireless car security system operating on the Global System for Mobile Communications (GSM) and Radio Frequency (RF) technology can remotely control and supervise your car. The available car security systems are almost under evaluation and currently they are not satisfying the requirements of security in sense of range, backup and cost. Even, time taken to get your money from insurance companies is a hectic job.

II. REVIEW OF THE STATE OF ART

Global System for Mobile Communications (GSM) security systems provide security in the form of remote arming/disarming, cutting off the lights, stopping car’s fuel or power supply, and locking or unlocking car door by telephone voice or by Short Messaging Service (SMS). Global System for Mobile Communications (GSM) available services like Short Messaging Service (SMS) is convenient and most popular way to communicate with each other. All networks provide Short Messaging Service (SMS) service is cheap flexible and a convenient way in converting data and this technology can be applied in many different areas like car security systems. The Global System for Mobile Communications (GSM), Global Positioning System (GPS) receiver and microcontroller can work together to give exact location of the vehicle. The microcontroller calculates and transmits the current location of Global Positioning System (GPS) system by means of Global Positioning System (GPS) data server [1].

In case of Radio Frequency (RF) systems, it is controlled completely by the car itself and there is no point of indulgence of the user. As soon as the stranger touches the car, siren sounds immediately which will follow signal lights flashing continuously? To turn off/on the siren and signal lights, the remote control unit is pressed once/twice while the alarm system is continuously and fully armed as shown in Figure 1.

Figure 1: Proposed Block Diagram
III. RESEARCH QUESTION AND PROBLEM STATEMENT

After long research, can range, backup systems, cost and location of a car be improved by effective switching between Global System for Mobile Communications (GSM) and Radio Frequency (RF) systems in a car security system?

IV. PROBLEM SOLUTION

For the first time, when the Global System for Mobile Communications (GSM) security system starts, the user gives 2 rings twice, to become an authorized person of the car.

Different Sensors are placed in the front and rear portion of car. Whenever a thief or a stranger touches the car on either sides of the car, sensors sense it and as a result, relay 1 becomes ON and hence microcontroller gets this message. The same type of message is received by microcontroller, even when car doors are touched when the relay 2 becomes ON. Microcontroller sends a message to the SIM module by converting 5V to 3V for which voltage controller is used. Now, SIM module sends message to the mobile of the owner of the car and requests for the proceeding action to be taken.

The message goes on repeating until car owner replies in the form of rings. As soon as he gives 4 rings to the SIM module, it again sends a message to the microcontroller through voltage converter converting 3V to 5V, which is the functioning voltage of microcontroller. When the Relay 3 is ON, automatically message is received by micro controller and it stops the functioning of the car. In all these processes external RAM is used to provide extra memory for the address to be addressed by the microcontroller.

When there is no signal poor signal the device will be unable to send the message in proper time frame, hence the car remains unprotected by this system. So in order to avoid this, Radio Frequency (RF) system which was bought from market in complete form, is used in car which will properly handshake with our Global System for Mobile Communications (GSM) security system with the help of microcontroller, by which our car gets protected. It is also connected with its own relay, microcontroller and internal car circuit.

For every 10 seconds microcontroller will check whether the signal quality of Global System for Mobile Communications (GSM) module is good or not. If it is poor than it will automatically make the Radio Frequency (RF) security system, the main security system of the car and also locks the functioning of the car with the help of relay connected to Radio Frequency (RF) security system. If after 10 seconds the signal quality of Global System for Mobile Communications (GSM) module gets good than it transfers the control of the security of the car to Global System for Mobile Communications (GSM) security system, otherwise it keeps the Radio Frequency (RF) security system as the main security system of the car.

The SIM module sends the user the location and status of the car with respect to area only when the user gives 3 rings for 1 time.

Moreover the SIM module will unlock the functioning of the car if the user gives 1 bell for 2 times to the module.

V. CONCLUSION

Complete integration of Radio Frequency (RF) and Global System for Mobile Communications (GSM) security system, overcoming the disturbances like shock, touching, breaking of cars window, door, bonnet sensors and ignition of car. The disabling of both the relays of Radio Frequency (RF) and Global System for Mobile Communications (GSM) system working with opposite functionality should be done in order to break through the car. A small SIM Module under Global System for Mobile Communications (GSM) system controls the whole car security with proper number of rings. Switching between Global System for Mobile Communications (GSM) and Radio Frequency (RF) is done instantly when one of the services goes down and vice versa with specific code which is known only to the user. In future this project can be enhanced by using Global Positioning System (GPS) operating mobile to give exact location of our location, a system which can burn the wiring of the car if it is confirmed that the car is stolen and satellite communication systems for better coverage and perfect location identification.

ACKNOWLEDGMENT

Many thanks to our families, especially our parents and Professor Wlodz Kulesza, without their encouragement and support we wouldn’t have been able to complete this uphill task successfully. We would also thank our batch fellows for sharing and conferring problems we faced during our work.

REFERENCES

Improving the SER of QPSK through LMS Adaptive Filter

Hamid Shahzad, Muhammad Imran Hasan and Asad Ali

Abstract— In this paper we proposed a solution to improve the SER performance of Quadrature Phase Shift Keying (QPSK) in a radio channel through adaptive filter. As there are several adaptive filters, it is better to choose the right one which is simple and has the ability to handle the signals properly. After studying adaptive filters of different types, we selected Least Mean Square (LMS) adaptive filter to improve the Symbol Error Rate (SER) of QPSK scheme. We analyzed that, by increasing the length of the training sequence and proper initialization of LMS adaptive filter algorithm, when a QPSK signal is passed through complex AWGN channel, SER of the received signal is improved.

Index Terms— Adaptive filter, QPSK, LMS

I. INTRODUCTION

This paper shows how to improve Symbol Error Rate (SER) of QPSK signal which passes through complex noise channel and LMS adaptive filter that leads to data comparison to take decision and plot the results. In QPSK half of the bandwidth is needed than the bandwidth required for Binary Phase Shift Keying (BPSK) in order to maintain the data rate. Our aim is to take the advantage of QPSK scheme over Binary Phase Shift Keying (BPSK) and using LMS adaptive filter algorithm to show the improvement of SER of the received signal. An analysis and demonstration of the graphs will be found at the end of this paper for better understanding.

II. REVIEW OF THE STATE OF ART

The LMS adaptive filter algorithm is the simplest and the most widely used because no matrices involved in it and it has the ability to adjust the weights of the filter tap and has low computational complexity with a stable behaviour when implemented in finite-precision arithmetic [1]. Modulation technique such as QPSK can fight much better than any other techniques against noise, keeping the same SER performance within the bandwidth and each symbol uses less spectrum to transport the information through a noisy channel [2].

When a QPSK signal passes through a channel, it faces blockades causing frequency amplitude and phase distortion affects the whole system’s performance. LMS algorithm based filter works well if proper initialization and training is performed while converging in very noisy and strong interferential environment. We can improve the performance of SER, by using training sequence which consists of single isolated pulse and is long enough so that inter-symbol interference totally diminishes between pulses [3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

A. Research Question

The research question is how to improve SER against inter-symbol interference, frequency and phase distortion in QPSK scheme by using LMS adaptive filter algorithm.

B. Problem Statement

The LMS adaptive filter algorithm can reduce the frequency and phase distortion when a signal passes through a complex Additive White Gaussian Noise (AWGN) channel. LMS adaptive filter’s stability and performance is better than other adaptive filters due to its robust algorithm.

In this research, we found the SER results with and without LMS adaptive filter and plotted the results through MATLAB simulation, these results are analyzed to find out whether LMS adaptive filter works well in QPSK scheme or not.

IV. PROBLEM SOLUTION

A. Method

A noise power of -10 to 10 dB is introduced in a complex AWGN radio channel and a QPSK signal with inter-symbol interference is transmitted through this channel. The received signal from the channel is carried via LMS adaptive filter. The LMS adaptive filter gets training sequence at the beginning of each block of data to train itself. On the basis of this training sequence the weight function or correction factor is calculated in order to reduce remaining errors in the received data from the radio channel. Once it has trained itself, there is no need for further estimation of data during the whole session of transmitted data sequence. The data with noise is then corrected on the basis of weight function.

The whole process is carried out by a simulation through MATLAB.

B. Simulation setup

We setup a simulation as depicted the Fig. 1. Here we used a Random data generator which feeds data to a converter to convert data to symbol and is transmitted to a complex channel with AWGN channel. We used LMS adaptive filter which gets data and training sequence to find the weight function or correction factor to estimate further errors in data
in order to improve the SER performance. After filtering, it forwards the data to data comparator and then to decision maker to plot the result.

![Fig. 1. Simulation Flow Chart](image1)

C. Results Analysis

The results found in Matlab are depicted in Fig. 2 & Fig. 3. At different noise power, the SER for QPSK signal is measured with and without LMS adaptive filter.

It is found that the output with LMS adaptive filter has low errors than the output that we got without adaptive filter, as shown in Fig. 2.

![Fig. 2. SER Plot Against Noise](image2)

If the numbers of training symbols are increased to train the LMS adaptive filter, the errors can be reduced in a great extent which is clearly depicted in Fig. 3.

![Fig. 3. SER Plot versus Training Sequence](image3)

V. CONCLUSION

An adaptive LMS filter, if well designed, effectivley can improve the SER performance in QPSK modulation technique. There are some other factors involved which carry certain weight to improve the performance of SER like, proper initializing and increasing training sequence.

In this research paper we trained the LMS adaptive filter once at the beginning of each short data blocks. In future, it is proposed to train the LMS adaptive filter on the basis of different time interval for a long data block.

ACKNOWLEDGMENT

The authors would like to thank Mr. Włodek Kulesza, professor at Blekinge Institute of Technology (BTH) for his continuing support to make this project, a success.

REFERENCES

Abstract: The transmission control protocol (TCP) is the most popular transport layer protocol on the internet, almost all the running applications over the internet relay on the TCP to ensure end-to-end reliable transmission. The degradation in the TCP performance due to the relatively high packet loss in the wireless LAN affect the work of the higher layer applications, such as normal web browsing and file transfer protocol. In this paper we study the performance of TCP based applications (web browsing) over WLAN 802.11, considering both the infrastructure and Ad-hoc architectures.

Index Terms— IEEE 802.11b, TCP performance, Quality of Experience (QoE).

I. INTRODUCTION

In recent years an increasing interest and demands in wireless internet access technologies for end subscriber have been shown by operators as a cheap alternative to traditional wired local loop. This direction stimulate a new concerns regarding the end user quality of experience (QoE), which will be effected by the performance of the underline internet stack protocols (most of the time the TCP protocol). The TCP based applications performance such as the file transfer and the normal HTTP traffic (web browsing) vary accordingly to the physical layer characteristics such as the delay and packets loss ratio.

Due to the ease of use and low cost the wireless network LAN components are available now almost in all new laptops, notebooks, and mobile phones. In addition to that the Wireless LAN access points are available on all the important public areas, airports, restaurants, offices etc. The 802.11 has two basic modes of operation; ad-hoc architecture enables peer-to-peer transmission between end user stations and infrastructure architecture in which the end user stations communicate through an access point that serves as a bridge to a wired network.

In this paper the end user Quality of Experience (QoE) has been examined in term of average response time of web browsing. The available WLAN models in OPNET simulator IT Guru academic edition 9.1 have been utilized to simulate the relation between the web average response time and different values for the WLAN frame fragmentation threshold (which simulate different values of transmission delay). More details and description of the simulation scenarios have been given in Problem Solution section.

II. REVIEW OF THE STATE OF ART

The TCP performance issues over the wireless networks and the resulted influence in the application performance have been discussed from many different aspects.

In [1] Wireless Web Performance has been discussed in a small class room conditions. The scope of work consist of an experiments on the HTTP transaction rate and end-to-end throughput achievable in the wireless network environment and the impacts of certain factors such as number of clients, Web object size and the connections persistency.

In [2] the impact of TCP sliding window on the performance of IEEE 802.11 WLAN are explained through the results of number of experimental scenarios of the contention between TCP loads and also shows the TCP unfairness through computer simulation and the interaction between the MAC and TCP transport layer mechanism.

In [3] an adaptive bandwidth sharing mechanism is used for improving and administrating the quality of service in infrastructure wireless networks.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Based on the previous study in TCP performance issues over wireless links [4]; the variation in the wireless link characteristics such as the packet loss ratio will cause variations in the TCP protocol performance over wireless network. The different WLAN network topologies (namely the Ad-hoc and infrastructure architecture) will have different links characteristics. So the TCP performance should vary accordingly. What is the effect of the TCP performance variation on the upper layer applications? The previous question can be mapped into more specific set of inquiries as follow:

1- What is the average variation in the page response time and the throughput from the end user /client point of view in Ad-hoc network when different value of fragmentation threshold is set?

2- What is the average variation in the page response time and the throughput from the end user /client point of view in infrastructure network when different values of fragmentation threshold are set?

Our main contribution in this paper is to estimate the average response time for the TCP based application over
different architectures of the WLAN.

Modeling the average response of web browsing and file transfer over WLAN is a challenge, due to fact that it depends upon number of factors, the capabilities of the used devices at communication ends in term of the CPU, RAM, processing load etc. In addition to the performance of the used protocols which can be described as follow:

\[ DT = C1 + C2 + P + Pe \times (C1 + C2 + P) \]

\[ DT \equiv \text{Total transmission time.} \]

\[ P \equiv \text{propagation time.} \]

\[ C1 = \text{processing time at the sender.} \]

\[ C2 = \text{Processing time at the receiver.} \]

\[ Pe = \text{Packet Error probability} \]

IV. PROBLEM SOLUTION

A. Simulation Steps descriptions

A simulation for WLAN has been implanted using OPNET IT Guru academic 9.1 for both infrastructure and ad-hoc architecture. Two main scenarios have been composed as follows: In the first scenario we simulate the operation of the small infrastructure WLAN; in which end user work stations are set at different distances from the wireless router to simulate different delay and SNR conditions. A heavy HTTP traffic (web browsing) has been generated between all the network clients and the Ethernet server. In the second scenario we simulate the operation of the small Ad-hoc WLAN. The end user work stations are set at different distances from the wireless server to simulate different delay and SNR conditions. A heavy HTTP traffic (web browsing) has been generated between all the network clients and the wireless server. The fragmentation threshold has been changed among ten different values and for each one of the set values the above two scenarios have been run.

B. Results layout

Graphs illustrate our simulation results have been plotted by MATLAB as shown in Fig.1 and Fig.2. The relation between the web browsing average response times and the fragmentation threshold for both infrastructure and ad-hoc has been illustrated in Fig.1; The X- axis represents the fragmentation threshold in bytes and the Y-axis represents the time in seconds.

![Fig. 1. Throughput comparison in ad-hoc & infrastructure architectures](image)

C. Results Analysis

From Fig.1 we can see that average response time is affected by smaller fragmentation threshold size (less than 200 bytes) for both cases; it decreases for the values between 256 bytes to 1056 bytes and keep decreasing beyond that value in the ad-hoc case. However the fragmentation threshold values greater than 1056 bytes caused rapid increment in the average response time. From Fig.2 we can see that the ad-hoc architecture shows maximum throughput above the mean value between 200 bytes and approximately 600bytes and also increasing behind the 1300 bytes while the throughput in the infrastructure have better values 256 bytes up to around 900 bytes (above the mean value).

V. CONCLUSION

Changing the WLAN parameters such as the fragmentation threshold values lead to an important change in the TCP based applications performance. From the results obtained in this paper we can conclude that the fragmentation threshold values between 256 bytes to 1000 bytes achieves better performance for the TCP based applications in both infrastructure and ad-hoc architecture.

In future this work can be extended to involve real network environment test scenarios and comparison can be performed against the simulation results. In this work more attention will be given to the obtained results in the case of ad-hoc when the fragmentation threshold is set above 1000 bytes.

REFERENCES


Abstract — VoIP Networks has blazed the trail as technology of choice for voice services in private and business establishments. With the increase in its usage comes an inevitable security concern inherited by the underlying IP network it rides on. In this paper we first mention other security solutions that are presently in use in the industry, and then we will focus on the implementation of an encryption standard the Advanced Encryption standard AES using Matlab programming. Finally we manipulated the encryption algorithm to suit our main aim of having good security yet with good speech quality.

Index Terms — Voice over IP, VoIP Security, Cipher Key, Shift Rows and Shift Column Transformation, State Matrix, AES.

I. INTRODUCTION

The recent increase in the usage of voice services over VoIP networks has opened up a host of security vulnerabilities, ARP Spoofing, eavesdropping, spam over etc. It has become imperative to encrypt our data payloads to ensure its integrity. One of several techniques in use is the AES which was introduced to deal with the drawbacks of Triple Data encryption standard 3DES and DES. The AES offers flexibility, security, high speed and low cost. Its flexibility allows to manipulate the algorithm to work with key and block size of any multiple of 32bits with minimum of 128bits and maximum 256bits.

II. REVIEW OF THE STATE OF ART

Potentially, every component of a VoIP system may be vulnerable starting from the physical layer through to the application layer. This has prompted the design of many solutions that follow closely the basic tenets of network security. Some designs focus on security on the physical layer while others go through the whole stack. An appealing solution in the defense against the denial of service DOS attack on availability is based on a good design of intrusion detection systems. According to an ISS whitepaper, “Intrusion Prevention is an essential element of a multi-layered VoIP security solution providing intrusion prevention at the gateway, on the network and at the host”[1]. The ISS method has capabilities of parsing and analyzing VoIP family protocols, including SIP, MGCP and SCCP. With these capabilities, it can preempt rogue traffic and hence create an alert. Another important defense mechanism used to protect against multiple attacks is port authentication [2]. Port Authentication requires a device connected to the network to authenticate itself with a central authority before it is granted access. Also we have established security design the creation of Virtual Local Area Networks VLAN’s used to separate data from voice traffic [3]. This implementation is carried out by switches which allow routing only from devices on the same VLAN. Another security line is with the SIP protocol which is used to connect VoIP phones in the network with the servers. When a phone registers with the SIP server, it provides its identity. This unique identity is made up of MAC and IP address of the phone. Precautionary measures are put in place to make sure that attacks such as MAC or IP address spoofing are limited. In this measure IPSec is used for authentication and encryption.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

The key security challenge is how do we guarantee security without a degradation of the speech quality? The problem of securing the voice packets leaving from one network to another is key to absolute security to most VoIP services. Therefore key cryptographic techniques have been used to encrypt the packets. The DES, IPSec, 3DES are all good examples. The problem here is that DES and IPSec are not secure enough while 3DES adds a lot of headers to the voice packets which affect speech quality. Our main contribution is that, we propose the implementation of the Advanced Encryption Standard AES with the modification of the algorithm to suit our desire goal of security with good speech quality.

IV. PROBLEM SOLUTION

The criteria we consider in the choice of our algorithm to offer a solution are listed as follows:

- Security
- Software & Hardware Performance
- Suitability in restricted-space environment
- Resistance to power analysis and other implementation attacks.

The trade-offs between choosing a computationally simple algorithm and desire to improve the call quality has a drawback in the sense that lighter algorithms are not secure. AES by design provides more security due to a larger block size and longer keys [4]. The unique thing with AES is that it could be manipulated to suit ones desired purpose. Finding a middle ground between call quality and security with choosing the right key length is a paramount design issue. In our unique

The Implementation of AES Algorithm using Matlab Programming for VoIP Networks

Emineimo Kennedy Obot Nwup, Naseer-Ud-Din Shinwari, and Riaz Hussain Junejo
solution we use the 128bit key. We modify the basics of the matrix formulations of the algorithm to make it more secure, flexible, efficient, implementable and proprietary.

The block diagram describing the basic AES algorithm is depicted below:

![Block Diagram of AES Algorithm](image)

Fig.1 Encryption Process.

AES is a key-iterated block cipher performing encryption and decryption for 128 bits input data and private key cipher which uses same key in encryption and decryption operations for 128, 192 and 256 bits [5]. In the encryption process, data to be encrypted is split into state matrix.

Round key is derived from the initial key and repeatedly applied to transform the block of plain text into cipher text blocks. The repeated application of a round transformation state depends on the block length and the key length. For various block length and key length variable’s value are given in table-1 [5].

<table>
<thead>
<tr>
<th>FLAVOUR</th>
<th>Block and Key Length Sizes</th>
</tr>
</thead>
<tbody>
<tr>
<td>AES-128</td>
<td>Key Length</td>
</tr>
<tr>
<td>4</td>
<td>4</td>
</tr>
<tr>
<td>AES-192</td>
<td>6</td>
</tr>
<tr>
<td>AES-256</td>
<td>8</td>
</tr>
</tbody>
</table>

Every round of the algorithm begins with “SubBytes” transformation. The S-Box is constructed by combining the inverse function with an invertible affine transformation to avoid the attacks.

ShiftRows is a transformation where each row of the state is shifted cyclically a certain number of steps. In the AddRoundKey step, each byte of the state is combined with the sub key. For each round, a sub key is derived from the main key using the key schedule.

Key Schedule: The sub keys are derived from the cipher key by means of key schedule with each round requiring Nb words of key date. The Round Keys consists of two components: the Key Expansion and the Round Key Selection.

We implemented AES using MATLAB. Here are initial State Matrix, Cipher Key, Round Key, Encrypted Matrix and the Decrypted Matrix.

\[
\begin{bmatrix}
00 & 0a & bc & 12 \\
aa & 3c & 10 & 27 \\
c1 & 22 & f1 & d2 \\
08 & 23 & 8d & b3 \\
\end{bmatrix}
\]

Input Matrix (Plain Text Matrix)

\[
\begin{bmatrix}
00 & 01 & 02 & 03 \\
04 & 05 & 06 & 07 \\
08 & 09 & 0a & 0b \\
0c & 0d & 0e & 0f \\
\end{bmatrix}
\]

Cipher Key Matrix

\[
\begin{bmatrix}
09 & bb & 59 & f9 \\
74 & c7 & 2d & 70 \\
ec & 64 & ea & e3 \\
3e & ee & e6 & a6 \\
\end{bmatrix}
\]

Output Matrix

V. CONCLUSION

AES by design provides more security due to larger block size and longer keys. The flexibility of its manipulation to suit one’s desired purpose makes it the algorithm of choice. The paramount design issue is to choose the right key length for security and call quality. For our solution the 128bit key length gave a good security level. In future the design issue of reducing the steps involved in the algorithm would make the design more simple, flexible, fast and implementable.

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Dynamic Spectrum Access in Cognitive Radio Networks
Aspects of Reactive and Proactive Sensing

Mohamed Hamid, and Subrata Kumar Das

Abstract—This paper considers the Medium Access Control (MAC) layer sensing schemes in cognitive radio networks, proactive and reactive sensing. Also in proactive sensing the adapted and non adapted sensing periods are both considered in this paper. All these schemes are assessed with two performance metrics spectrum efficiency and idle channel search delay. The simulated cognitive radio network is an ad-hoc network supporting data transfer among its nodes. Simulation results completely match the theoretical ones.

Index Terms— Spectrum Sensing, Spectrum efficiency, idle channel search delay, Spectrum holes.

I. INTRODUCTION

Dynamic spectrum access is one of the most precious features in cognitive radio networks as it aims to utilize the available spectrum dynamically to solve the problem of congested bands yielding high utilization of the radio resources. The main motivation of this paper is the need of finding MAC layer access scheme to the unlicensed users in cognitive radio networks.

In this paper we focused on MAC layer sensing scheme as MAC layer is the most important layer to be considered in promising radio networks.

II. REVIEW OF THE STATE OF ART

In the last decades notions about radio have been changed and combination of hardware and software based radios are evolved. In the early 1990's Joseph Mitola introduced the idea of software defined radios and in 2000 he introduced the idea of cognitive radio. Recently working on MAC layer spectrum sensing in cognitive radio networks became a hot area of research and one of these researches was done by Hyoil Kim and Kang G. Shin in University of Michigan and they proposed proactive sensing as a new sensing scheme in cognitive radio networks MAC layer [1].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

A. Background

Existing way of spectrum management relays on a static spectrum allocation policy, that is each licensed spectrum band is statically assigned to the specific licensed service and its users. However, the huge increase in technologies and services based on radio communications resulted in serious need of dynamic spectrum allocation to overcome the static allocation limitations.

B. Research focusing

The main Idea of this paper is to investigate the MAC layer dynamic spectrum access schemes for unlicensed users to detect the available spectrum holes and to asses each access scheme using some performance metrics. Thus our main task is to determine how the unlicensed users should sense the spectrum and in which pattern they have to do this in order to achieve as high spectrum efficiency and as low idle channel search delay as possible.

The unlicensed users can sense the spectrum either in a reactive way or in a proactive one [1]. To estimate the performance of each type of the pre-mentioned types of sensing two performance metrics were taken into account those are idle channel search delay and the spectrum efficiency. Idle channel search delay is the time the unlicensed user needs to discover an idle channel that can be utilized while the spectrum efficiency is the portion of the available spectrum that the unlicensed user can utilize sending and receiving data.

This paper contributes in cognitive radio area in the following means:

* Providing one way for the unlicensed users to select one of two available sensing schemes.
* Propose and assess sensing periods adaptation method in order to achieve as high spectrum utilization as possible and compare it with the non adaptive one.

IV. PROBLEM SOLUTION

A. Channel Usage model and channel parameters estimation:

In order to simulate and asses the performance metrics some assumption are considered. At first the simulated cognitive radio network is an Ad-hoc network supporting data communication between its nodes. Another important assumption is that the available channels can be modeled as ON/OFF channels or 0/1 states: 0 for free channel and 1 for occupied channel which is referred to as channel usage model as illustrated in Fig.1.
Fig.1 ON/OFF Channel Usage Model

We also assumed that the unlicensed user always has a packet to be sent or received which yields that the free channels are always needed to be utilized. One more important assumption is regarding the unlicensed user node equipments as it should be supported by a wideband tunable antenna which can work on the whole available spectrum and perform one task at a time either sensing channels or communicating with another node.

In proactive sensing we have two additional factors, the first is the unexplored opportunities of free channel due to discrete sensing and the second is the sensing overhead and the meaning of that is no other unlicensed user can utilize the channel during the sensing procedure of any unlicensed user. However these two factors should be traded off with reducing the idle channel search delay.

The number of channels assumed to be N each is addressed as channel $i$ where $i=1,2,\ldots,N$ and they are assumed to have an exponential distribution of both ON and OFF periods with means $E_{\text{ON}}$ and $E_{\text{OFF}}$ respectively, and from the distribution parameters the unexplored opportunities and sensing overhead can be determined using renewal theory [2].

B. Obtained Results and their interpretation:

MATLAB based simulation has been implemented to investigate reactive and proactive sensing and adaptive sensing periods proactive sensing versus non adaptive one as well and for that we hypothesize 5 channels with a specific different ON/OFF means as follows $E_{\text{OFF}} = 25, 35, 4, 50$ and 5 seconds respectively $E_{\text{ON}} = 5, 15, 20, 30$ and 5 seconds respectively. Also we set the time needed to sense each channel to 20ms.

A. Channel Sensing Period

Fig.2 Sensing Periods and their corresponding performance in terms of wasted spectrum in sensing procedure of channel 2.

Fig.2 demonstrates the optimum sensing period for channel 2 which has less than 10% sensing overhead and unexplored opportunities.

B. Performance of the reactive and proactive sensing

Fig.3 Proactive and Reactive Sensing idle channel search delay versus number of channels

It can be figured out from Fig3 that proactive sensing is much better in this sense of idle channel search delay when the number of channels increase.

C. Performance of the adaptive and non adaptive sensing

Fig.4 Spectrum efficiency of the adaptive and non adaptive sensing with the Elapsed Simulation Time

As in Fig.4 adaptive proactive sensing has consistent spectrum efficiency with the time while non adaptive one has not.

V. Conclusion

The pre-illustrated results conclude that to achieve as high spectrum efficiency and as low idle channel search delay as possible the proactive sensing with adapted sensing periods is the best solution for that.

For the future work, studying the impact of the sensing schemes on nodes complicity and the trades off in that is highly recommended.

REFERENCES


Design of Microstrip Antenna with Maximum Bandwidth for IEEE 802.16-2004

Tulha M. Yazdani, Gu L. Feng, and Arif Hussain

Abstract—The paper aims to design a microstrip antenna (rectangular patch antenna) with suitable height of dielectric substrate (air) to achieve maximum bandwidth for IEEE 802.16-2004 standard. The microstrip antenna is designed for 3.5 GHz resonant frequency. The transmission line model is used as a method of analysis. The coaxial feed is selected as a feeding method. The MATLAB program is used to observe the bandwidth response at different height. Different substrate heights are selected using iteration method defined in paper. The bandwidth response of 75 MHz is obtained with substrate height 4mm using MATLAB program. The microstrip antenna is fabricated using Printed Circuit Board (PCB), Bayonet Neill Concelman (BNC) connector, coaxial cable. The fabricated antenna with substrate height 4mm is subjected to spectrum analyzer gives 69 MHz bandwidth response. The variation from desired result (75 MHz) is subjected to lack of accuracy in fabrication. The designed antenna is suitable for worldwide interoperability for microwave access (WiMAX) application based on IEEE 802.16-2004.

Index Terms—Microstrip Antenna, Transmission Line model, Dielectric Substrate

I. INTRODUCTION

Nowadays microstrip antennas are very popular among Local Area Network (LAN), Metropolitan area Network (MAN), wide Area Network (WAN) technologies due to their advantages such as high operating frequency, impedance matching, compatibility with integrated circuit, light weight, low volume, inexpensive.

The main disadvantage of microstrip antenna is its narrow bandwidth. During the last decade techniques (Increase in substrate height, use of electrically thick element) are developed to increase the bandwidth of microstrip antenna. Variable heights (using iteration method defined in this paper), frequency (3.5 GHz) and dielectric constant (σ=1) are input to a MATLAB program to observe the bandwidth response. The desired 75 MHz bandwidth is obtained at 4mm.

To verify results the microstrip antenna is fabricated at substrate height 4mm and subjected to spectrum analyzer gives 69MHz bandwidth response. The variation in result is due to lack of accuracy in fabrication.

This paper leads to broadband and low cost microstrip antenna suitable for worldwide interoperability for microwave access (WiMAX) application based on IEEE 802.16-2004.

II. REVIEW OF THE STATE OF ART

Paper [1] talks about design of broadband microstrip antenna using low-high-low (sandwich) dielectric constant.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Select the suitable height of substrate of microstrip antenna in order to get maximum bandwidth for WiMAX application based on IEEE 802.16-2004.

The design leads to wide bandwidth and low cost microstrip antenna for WiMAX application based on IEEE 802.16-2004. Increasing network cost in WiMAX technology is major drawback.

IV. PROBLEM SOLUTION

A. Iteration Method

The iteration method to select a suitable height (h) of dielectric substrate of microstrip antenna to be bandwidth compatible to IEEE 802.16-2004 standard is as under.

According to theory the height (h) of a substrate of microstrip antenna is defined between the ranges \(0.003 \leq h \leq 0.005\). Let height \(h_1\) define the lower limit, height \(h_2\) define the upper limit and height \(h_3\) define the center point of the span. This divides the whole span in to two parts. The span from \(h_1\) to \(h_3\) is named region A. The span from \(h_2\) to \(h_3\) is named region B. Observe the bandwidth response at \(h_1\), \(h_2\), \(h_3\). If the desired result is not obtained then take a middle point between \(h_1\) to \(h_3\) name it \(h_4\) also take a middle point between \(h_2\) to \(h_3\) name it \(h_5\). Check the bandwidth response at \(h_4\) and \(h_5\) as if you get the desired result. If not, compare the bandwidth response at \(h_4\) and \(h_5\). Select the height with better bandwidth response. Let \(h_4\) gives a better bandwidth response than \(h_5\). So now you will work on region A (\(h_1\) to \(h_3\)). Repeat the same procedure until you reach the desired result. The accuracy depends upon number of iteration.

B. Design

The microstrip antenna (rectangular patch antenna) is designed [2] and fabricated. The transmission line model is selected as method of analysis because it is simplest of all, gives good physical insight and easy to fabricate. Coaxial feed is selected as feeding technique because it is the most common
technique in practical model, easy to fabricate and easy to match.

Impedance is different at different location with its peak value at edges of patch (150 to 300Ω) and reduces to 0Ω. The feed point is positioned at third way along radiator (X=1.3mm and Y=0.06mm) match to 50Ω.

The ground plane can be infinite but due to physical constraints this is not possible. A reasonable square pattern of 120mm ground plane is selected.

3.5 GHz frequency is selected within 2 to 11 GHz band for IEEE 802.16-2004. Air is selected as a dielectric substrate with dielectric constant ε=1 because if air undergoes a dielectric breakdown the breakdown is not permanent whereas in solid dielectric substrate e.g. glass, polyethylene etc can sustain permanent breakdown.

Variable heights, frequency 3.5 GHz and air dielectric constant ε=1 is input to a MATLAB program to observe the bandwidth response and radiating patch dimensions.

![Fig. 2. Shows impedance match at 51.09Ω which is close to 50Ω as desired](image)

The heights are selected on the basis of iteration method defined above. Graph shows that 75 MHz bandwidth is achieved at substrate height 4mm. The dimension of radiating patch is found to be 37×37mm (L×W).

C. Fabrication

The fabrication process of microstrip antenna is explained as Select a PCB board with copper on one side and plastic on other. Cut a PCB according to dimension of ground plane and radiating patch. Drill a tiny hole at feed point location. Fix a female BNC connector to center of ground plane. The inner conductor of coaxial cable extends through dielectric and is soldered to radiating patch at feed point location. The other end of conductor is connected to female BNC connector.

![Fig. 3. Shows the bandwidth of antenna is 69MHz. The resonant frequency is 3.04 GHz instead of 3.5 GHz](image)

D. Validation

In order to verify the bandwidth response at 4mm substrate height obtained from simulation result the fabricated microstrip antenna was attached to spectrum analyzer through a coaxial cable jumper. Impedance matching and bandwidth response graph is shown in fig2 and fig3 respectively.

![Fig. 1. Shows the Bandwidth Response at different heights](image)

The result shows that fabricated microstrip antenna is matched at 51.09Ω and resonant at 3.04 GHz gives 69 MHz bandwidth. The variation from simulated result is due to mismatching, change in resonant frequency, inductance capacitance effect of BNC, slight variation in dimension while cutting and minute change in substrate height ± 5%. Based on these result we conclude if design is precisely followed the practical microstrip antenna will meet the simulation result.

V. Conclusion

This design leads to wider bandwidth and low cost microstrip antenna for WiMAX application. Design of broadband microstrip antenna for portable devices will be a great contribution in technology

REFERENCES


Comparison of the bit error rate (BER) of simple CDMA and Rake Receivers using Matlab

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Abstract— Code division multiple access (CDMA) technology provides high data rate as compared to Global system for mobile communications (GSM). In this research we implement simple CDMA transmitter, receiver and rake receiver. Then simulate the Bit error rate (BER) of both the receivers and compare the results of our experiment to provide the receiver with less BER in the received signal.

Index Terms— Bit error rate (BER), Line of sight (LOS), Code division multiple access (CDMA).

I. INTRODUCTION

Today we live in 3rd generation era and the technology became fast, data rate became very high as well as application requirements are very high. The service providers must meet the promised quality of service. CDMA technology provides high data rate compared to GSM. It is suitable for applications that require high data rate. In CDMA system a Rake receiver is use. A Rake receiver receives signals on as many multipaths as possible and then combines them to produce one clear signal that is stronger than the individual multipath. The Rake receiver attempts to combine the energy of several multipaths in order to maximize the energy of the received signal.

This research is all about reducing BER in the received signal. We use random signal to transmit and receive that is simple CDMA transmitter and receiver and then calculate BER. Further we use three signals which consist of line of sight (LOS) signal as well as non line of sight (NLOS) signals and these signals will be received at three finger rake receiver. Then we finally calculate the BER of this experiment and analyze the error rate of both the receivers.

II. REVIEW OF THE STATE OF ART

A large number of results can be found on Implementing simple CDMA transmitter and receiver but the study is limited to implementing Direct-sequence spread spectrum (DSSS)/CDMA transceiver [1]. A Rake structure is used to resolve the time diversity due to the multipath propagation [2].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

After this research we are able to answer the following questions.

- How to implement simple CDMA transmitter, receiver and Rake Receiver using matlab?
- Which receiver is best in BER perspective?

IV. PROBLEM SOLUTION

A. Implementation of CDMA Transmitter:

In order to transmit the data we used the following functions. These functions together form a transmitter.

- Input data: In our experiment 60 bits of data are randomly selected and applied to the transmitter as input.
- Spreading: The input data is spread with Walsh code. Walsh code is generated by hadamard matrix which is actually a square matrix whose entities can be only +1 or -1. The Walsh code of $2^n$ length can be used maximum up to 512 and we have used Walsh code of length $2^4 =4$ in our simulation. In order to generate a Walsh code that spreads our data eventually, we have made a function in matlab.
- Scrambling: The spreaded data is then scrambled by PN code. In CDMA system each user is assigned a unique PN code. PN Code (Pseudonoise) is generated by multistage shift register, where some selected outputs are added modulo 2 and then feedback to the input of the multistage shift register. The scrambling is done by a function in matlab.
- Modulation: We used BPSK Modulation to modulate the Scrambled data. BPSK Modulation is built-in function in Matlab.

B. Implementation of CDMA Receiver:

Similarly following functions form a complete CDMA receiver.

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- Demodulation: This function is used to demodulate the data modulated in the transmitter. This is built-in function of Matlab.

- Descrambling: After demodulation the data is then descrambled by removing PN code. The descrambling process is simply the XOR (Exclusive OR) operation of demodulated data with the PN code that is the same used at the transmitter. The function is made for this purpose.

- Walsh decoding: After descrambling the data, Walsh decoding is applied. The Walsh decoding process is simply the XOR operation of the descrambled data with the Walsh code that is the same used at the transmitter. The function is made for this purpose.

- Despreading: In this function we use integrator, the decision threshold decides, based on the output of the integrator. If the output of the integrator is greater than 0 the output is 1 else the output is 0.

C. Implementation of rake receiver

Our simulated rake receiver consists of three fingers that are capable of receiving three strongest multipaths.

- First multipath is arriving with no delay (LOS).
- Second multipath is arriving at a delay of one chip.
- Third multipath is arriving at a delay of two chips.

The following functions are used for CDMA rake receiver

- Demodulation: Demodulation is carried out using built-in function in matlab.

- Descrambling: The main difference between de-scrambling process for the Simple CDMA receiver and the Rake receiver is the later de-scrambles multiple replicas of the received signal which have been delayed by pre-determined factors. In each finger, the signal is descrambled by the PN sequence used at the transmitter. However the delays are also incorporated in the descrambling process.

- Walsh decoding: This function work similarly as in the simple CDMA Receiver.

- Diversity Combiner: After decoding the data with Walsh code, data from each of the three fingers are added constructively to get a stronger signal.

- Integrator circuit decision threshold: The output of the diversity combiner is then fed to an integrator circuit decision threshold, if the output of the integrator is greater than 0, then the decision is 1 else decision is 0.

D. BER of CDMA and Rake receivers

Simulation Result:

Simple receiver:

\[
\text{simple\_rec\_error} = 166.6667 \\
\text{Simple\_BER} = 0.1667
\]

Rake receiver:

\[
\text{rake\_rec\_error} = 83.3333 \\
\text{rake\_BER} = 0.0833
\]

V. CONCLUSION

In this research simple CDMA transceiver and rake receiver are implemented in Matlab. The BER response is checked for both simple and rake receiver. It is concluded from these experiments that BER of Rake receiver is much lower as compared to simple CDMA Receiver.

We considered the BER in CDMA; the work can be extended to find the effect of using multiple signals on the power, delay and other factors.

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REFERENCES


Abstract—This paper presents the effect of Multiple Access Interference (MAI) on bit-error-rate (BER) performance of a direct sequence photonic code division multiple access (DS-OCDMA) network based on star coupler with intensity modulation and sequence inversion keyed (SIK) optical receiver correlator. The BER versus Received optical power performance results are evaluated with 7 chip m-sequence considering the effect of MAI using Matlab simulation. It is found that due to MAI effect the Bit-Error-Rate is degraded when the number of users and received optical power is increased. From simulation results it is observed that the performance of BER can be developed by adding the chip numbers per bit.

Index Terms— Code division multiple access (OCDMA), Multiple Access interference (MAI), Sequence Inverse Keying (SIK).

I. INTRODUCTION

At present OCDMA technique is vastly studied to ensure the high performance of the transmission because of its several attractive features such as asynchronous access, privacy and security in transmission, ability to support variable bit rate and busy traffic and scalability of the network [3]. Intensity modulation and direct detection on-off keying (OOK) OCDM and pulse position modulation (PPM) OCDMA are already analyzed [1]. The capacity of these networks is limited because the number of signature sequences available with good correlation properties for a given sequence length is small [2]. Referenced above, all of these papers are on OCDMA network, performance analyses are found out without considering the m signature sequence. That’s why MAI results degrade the Bit-Error-Rate performance in the OCDMA system [2]. In this paper an analytical expression as well as Matlab simulation is presented to calculate the Bit-Error-Rate of OCDMA network based on star coupler. In this thesis we have simulated on BER versus received optical power of an OCDMA transmission system for various numbers of users with 7 chip m sequence when the chip sequence is 1110010.

II. REVIEW OF THE STATE OF ART

In the [2] we found that the paper is analyzed on receiver section. But OCDMA consists of Transmitter, Channel and Receiver parts. We found that the paper used the PN signature sequence. We can improve the performance of the system using m-signature sequence, gold signature sequence.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Our research questions are as follows
- Which method is used to determine the MAI in the OCDMA system?
- How can we validate the method in Matlab toolbox?

The Bit-Error-Rate is degraded when the number of users and received optical power is increased due to the MAI in OCDMA system.

The contributions that we have made in this paper are written.
- To find out the Multi-Access Interference (MAI).
- To find out the signal to MAI ratio.
- Finally, to find out the BER performance of an OCDMA system.

IV. PROBLEM SOLUTION

A. System Description

A schematic block diagram of an Optical Code Division Multiple Access communication system with Figure 1(a) Optical Transmitter and (b) SIK Switched Correlator Optical Receiver [1] are drawn.

In the Optical transmitter section, a user’s binary data is modulated either by a unipolar signature sequence or by its complement depending on whether it is a “1” or “0”, respectively. Now the signal is encoded. Using K: 1 coupler the encoded signals of K number of users are coupled and this signal is transmitted through an optical transmission line. In the Optical Receiver section, in order to get the original data, directly a bipolar reference sequence is correlated with the channel.
unipolar signature sequence. The bipolar reference sequence is separated into two complementary unipolar reference sequences by means of unipolar switching functions, to despread the incoming optical channel signal [3]. Finally a balanced p-i-n photodiode subtracted the despreaded optical signal and integrated this signal over a data bit period before to detect the zero thresholds.

B. System Analysis

The total received Optical Signal with SIK Correlator in the DS-OCDMA; \( R(t) \) is given by [2]

\[
R(t) = \sum_{k=1}^{K} P_k B_k(t-\tau_k) \otimes A_k(t-\tau_k) \quad \quad (1)
\]

Where \( P_k \) is the chip optical power for the K-th user \( B_k(t) \) and \( A_k(t) \) are binary user’s data and signature sequences respectively. The time delay is \( \tau_k \). \( K \) is the simultaneous number of users. The operator \( \otimes \) is the SIK modulation so that either the sequence \( A_k(t) \) or its complement \( \bar{A}_k(t) \) is transmitted for a data bit ‘1’ or ‘0’ respectively. Sequence of unit amplitude unipolar rectangular pulses of duration \( T \) is \( B_k(t) \). A periodic sequence of \( N \) unit amplitude unipolar rectangular pulses of duration \( T_c \) is \( A_k(t) \). Thus, can be written as \( T \sum_{k=1}^{K} \frac{R(t)}{N} \). The Correlator Receiver output signal \( Z_i(t) \) matched to the \( i \)th user is [2]

\[
Z_i(t) = T \sum_{k=1}^{K} \frac{R(t)}{N} \left[ \sum_{l=0}^{N-1} \left( \sum_{l=0}^{N-1} \frac{R(t)}{N} b_i(T_r - \tau_l) \alpha_i(T_l - \tau_l) \alpha_i(T_r) \right) dt \right] Z_i(t) dt \quad \quad (2)
\]

Where \( l \) is the \( h \)th chip delay, \( R \) is the responsivity of the photodiode and \( M \) is the mean avalanche multiplication factor. \( G \) is the gain of optical amplifier. \( \bar{A}_i \) is the complement of \( A_i \). The total channel noise at the correlator output is \( n_o \). Here, \( a_i(t) = \{ A_i(t) - \bar{A}_i(t) \} \), where \( a_i(t) \) is the bipolar form of the signature sequence, and \( B_i(t) \otimes A_i(t) = (1 + b_i(t)\alpha_i(t)) / 2 \). where \( b_i(t) \) the bipolar form of the data sequence [2]. In the above (2) the first term is mean (U), the second term is variance of MAI (\( \sigma \)) and the remaining is the Noise (\( n_o \)).

\[
\sigma^2 = \left[ T \sum_{k=1}^{K} \frac{R(t)}{N} \right] \left[ \frac{2(k-1)}{3N} \right] \quad \quad (3)
\]

Where, \( q \) is the electronic charge.

\[
U = \frac{T \sum_{k=1}^{K} \frac{R(t)}{N}}{q} \quad \quad (4)
\]

At the correlator output, the signal to noise is found as

\[
SNR = U^2 / (\sigma^2 + n_o) \quad \quad (5)
\]

The BER for OCDMA transmission system is then given by [1].

\[
BER = (1/2) erfc\left( \frac{\sqrt{SNR}}{\sqrt{2}} \right) \quad \quad (6)
\]

C. System Performance

The following figure 2 shows the plots of Bit Error Rate versus Received Optical Power for more than one user, 7 chip m-sequences. It is found that the BER decreases with the increase in Received Optical Power. For BER = 10^-10 the required Received Power is -15dBm.

![Figure 2 Plots of BER versus received optical power (p_s) for various numbers of users with 7 chip m-sequences](image)

V. CONCLUSION

The accomplishment of this thesis is the result of great effort of group members and the full devotion of our teacher. We are able to complete this thesis with the prayers of our parents.

REFERENCES


Abstract—Importance of training a network with Independent Component Analysis (ICA) arises in Blind Signal Separation (BSS) especially when there are strong background speech signals present in the environment. In blind signal separation, the goal is to recover all original source signals by only using the observed mixtures. In this paper, ICA Algorithm is used to test the performance of a trained network when different mixture patterns are introduced. Simulation results prior to training and after training are illustrated for further analysis. As a result, training the network with multiple patterns with a small, well predicted and bounded gamma value provides lower root mean square error in the estimated signal with respect to the original signal and a better estimation performance with trained neurons in real environment.

Index Terms—Neural Networks, Signal Detection, Error Analysis.

I. INTRODUCTION

ICA is a widely used unsupervised algorithm that can separate each source signal within a mixture of signals by using observation principle [1]. Without prior knowledge about the source signals, ICA is able to separate signals [2]. There are various kinds of ICA that are chosen to be used for different applications. To choose one ICA over another depends on the performance of the correlation between estimated signals and original sources. In addition to that, the performance of signal detection and separating each statistically independent source signal can be improved by training the network with many different patterns. Patterns are referred to as elements of a set. When a network is trained to a single pattern, the corresponding accurate output signal is observed and however when a new pattern is presented to the same trained network, the network would not be able to produce the expected accurate separated signal. This problem can be solved by training the network with multiple patterns [3]. Error analysis made for the simulated experiment shows that the Root Mean Square Error (RMSE) decreases with increase in the number of different training patterns introduced to the network. The selection of the gamma value within a bounded range helps to obtain the estimated signal which is closer to the original signal.

II. REVIEW OF THE STATE OF ART

To consider which ICA algorithm to use in the process depends on the application that is being worked on. However mainly the algorithm is as follows [4]:

1. Choose an initial random weight vector \( w \).
2. Calculate the \( y(n) = W \times x(n) \)
3. Calculate the weight vector:
   \[
   W_{\text{new}} = W_{\text{old}} + \{\text{gamma} \times [(W^T)I - (\frac{2}{\lambda}) \times x^T \times \tanh (y)]\}
   \] (1)
4. If not converged, go back to 3.

Using ICA, vector \( x \) is obtained by a transformation matrix \( A \) (random weight vector) and vector \( s \) (source signals); where it is modeled as \( x = As \). Thus, \( s = [s_1, s_2, s_3]^T \) is linearly mixed with \( A \) to obtain \( x = [x_1, x_2, x_3]^T \). We used \( y(n) = W^sx(n) \) for finding the estimated source matrices after applying the iteration of ICA algorithm; where \( W \) denotes the de-mixing matrix [5]. By choosing appropriate gamma and frequency value in the simulation, better signal can be obtained. The gamma has a bounded value between zero and \( 2/\lambda \), where \( \lambda \) is the maximum eigen value of the autocorrelation of \( x \).

The root mean square error between the estimated and original signal is calculated with the following formula:

\[
\varepsilon_{\text{rms}} = \left[ \frac{1}{MN} \sum_{x=0}^{M-1} \sum_{y=0}^{N-1} [f(x,y) - f(x,y)]^2 \right]^{1/2}
\] (2)

III. RESEARCH QUESTION AND PROBLEM STATEMENT

In this experiment we intend to analyze ‘Dependence of the RMSE on the Number of Training Patterns and Gamma for ICA Algorithm’. We focused on ICA and approached the problem to justify ‘The performance of the trained network increases with the number of times the network is trained with different patterns using a well predicted small gamma value’. We are trying to train neurons as much as possible to obtain better results. Our main statement is supported by a Matlab simulation as can be seen in the problem solution, Fig. 2 and Fig. 3.
IV. PROBLEM SOLUTION

At first, the potency of the ICA algorithm is tested with just one set of mixture signal, as in Fig 1. After obtaining a satisfactory correlation between the estimated and the original signal, the trained network is applied to various kinds of mixture sources. For every newly introduced pattern, one unique neuron will be more experienced to recognize new un-introduced patterns. Thus, more experienced neurons will be obtained if more training patterns are introduced to the network. Therefore, the potency of the algorithm and process of recognizing new patterns will be rapidly improved.

During the simulation, we used three different music signals as shown in the first column in Fig 2. All three signals are mixed with each other as shown in the second column in Fig. 2 and are separated with the previously stated equation and algorithm. The output for the separated mixtures is shown with the third column in Fig. 2. It can be seen from the simulation results that they are almost equal to the original signals and we conclude that the algorithm works fine for these patterns.

The network is trained with the same algorithm but with many different patterns. It’s clearly recognized that the RMSE decreases exponentially as the number of trainings of the network increases as shown in Fig. 3. In addition, selection of appropriate gamma values for ICA within the bounded range helps the network to result in lower RMSE errors, as shown in the second and third rows of Fig. 3. It’s clearly seen that the network behavior is not linear and not bounded if a high gamma value is selected over the boundaries, first row of Fig. 3.

V. CONCLUSION

We presented the aided performance of introducing different patterns with small gamma value on the trained neural networks. The validity of the hypothesis is shown by simulations. The accuracy we obtained from the results show that the potency of the trained network increases proportionally with the number of newly introduced training patterns. Choosing more accurate smaller gamma value for ICA helps the network gain a linear and bounded characteristic behavior with less RMSE.

For the improvements of this paper, a Matlab simulator can be designed which calculates the minimum number of patterns required for training and gamma value to find the tolerated RMSE. However, the processing time for calculating the iterated new weighted vector is a problem. By using different preprocessing (centering, whitening) prior to ICA calculation may be helpful to speed up the execution of simulation. Applications of this experiment can lead to improvement not only in audio processing but also in biomedical signal processing and telecommunications.

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REFERENCES

Simulation of BER vs. $E_b/N_o$ for AWGN and Rayleigh Channel using BPSK Modulation

Muhammad Umair Aslam, Taimoor Siddique, and Muhammad Haris

Abstract—Binary-Phase Shift Keying, BPSK is one of the most effective techniques of modulation. Whenever we want to maintain a lower value of Bit Error Rate, BER for different channels, we have to compromise remarkably over Energy per bit to Noise Power Spectral Density Ratio, $E_b/N_o$, which also gets affected adversely due to the effort and vice versa. This research explains that the most favorable condition is possible to achieve for Rayleigh Fading Channel modulated in BPSK to have better and suitable $E_b/N_o$ value for same BER as compare to Additive White Gaussian Noise, AWGN Channel. An algorithm is devised and MATLAB simulation results endorse the suitability for practical implementations.

Index Terms—Fading channels, Phase shift keying, Rayleigh channels.

I. INTRODUCTION

In the recent years, increasing attention has been paid to the modulation techniques. This paper draws upon the analysis of Rayleigh Fading Channel using Binary Phase Shift Keying, BPSK and Additive White Gaussian Noise, AWGN Channel and to provide a considerable attention to the relationship between Energy per Bit to Noise Power Spectral Density Ratio, $E_b/N_o$ and Bit Error Rate, BER using MATLAB implementation. This work motivates the new researcher to observe the simulated results. Our main results show that by using BPSK modulation for Rayleigh Fading Channels and obtaining its simulation in MATLAB we can achieve a suitable scenario for the channel’s BER vs. $E_b/N_o$ characteristics.

II. REVIEW OF THE STATE OF ART

The basics and constructional work of BPSK modulator provides the foundational knowledge [1]. From the derivation of the average BER of BPSK showing good performance when there is no phase error gives theoretical approach [2] while BPSK offers acceptable BER during transmitting signal of low energy or power [3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Normally in the receiving end SNR is very low because of Rayleigh fading and for this purpose we do modulation. Our research question is to find the relationship between BER and $E_b/N_o$ with AWGN and Rayleigh Fading Channel using BPSK and to simulate in MATLAB.

The contribution of this paper is that we show the implementation of BPSK algorithm in MATLAB for describing the channel’s BER vs. $E_b/N_o$ characteristics in the two different cases.

IV. PROBLEM SOLUTION

From the bit generator a random binary sequence consisting of 1s and 0s is generated in such a way to detect low signals [1] and a phase shift of 180° is applied in accordance with digital stream [1] for BPSK.

From the following flow chart given in Fig.1 and Fig.2, we will get the simulated Fig.3 and Fig.4.

Fig. 1. Flow Chart of BPSK transmitter-receiver with AWGN.
We can clearly demonstrate from the MATLAB simulated results obtained in Fig. 3 and Fig. 4 that when modulation scheme BPSK is employed for the same BER we have attained higher value of $E_b/N_0$ by using Rayleigh Fading Channel with respect to AWGN channel.

Fig. 2. Flow Chart of BPSK transmitter-receiver with Rayleigh Fading Channel.

Fig. 3. BER curve for BPSK Modulation in AWGN in terms of $E_b/N_0$.

Fig. 4. BER curve for BPSK Modulation in Rayleigh Fading Channel in terms of $E_b/N_0$.

$E_b/N_0$ characteristics of the BPSK modulation for Rayleigh Fading Channels approximates around the value 40 while from AWGN it stops before 10 of the x-axis.

V. CONCLUSION

We have shown the implementation in MATLAB for explaining the channel's BER vs. $E_b/N_0$ characteristics in the AWGN and Rayleigh Fading channel model using BPSK modulation.

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REFERENCES


Fetal ECG extraction by Adaptive Noise Cancellation with LMS and RLS algorithms: A comparative study

Iftekhar A. Shaikh, Mozghan Hedayati and Ramaz Shmhan

Abstract—In this paper we have made an effort to demonstrate the usefulness of Adaptive Filtering approach to extract Fetal electrocardiogram (FECG) from Maternal electrocardiogram (MECG) as well as a comparative study is done of the approach by the use of Least mean square (LMS) and Recursive least squares (RLS) algorithms. Maternal abdominal ECG (MECG) contains information about fetal heart rate, but that information is contaminated. Fetal heart rate information is vital in determination of heart anomalies, presence of multiple fetuses etc. Identification of these problems in early pregnancy can reduces risks by timely treatment or planned delivery. Based upon any of these algorithms a medical information module, as a part of Medical information systems could be develop.

Index Terms—Adaptive signal processing, Electrocardiography, Medical information systems.

I. INTRODUCTION

THE extraction of FECG is very vital for the early detection of ailments and defects in fetus. This early detection can assure good health and safety of fetus.

Some specific sources of interference arise in the detection of the ante partum FECG from electrodes on the mother’s skin. The omnipresent Maternal ECG (MECG), which can be 5-1000 times larger in its intensity, forms the largest interference. While Electromyographic (EMG) activity, potential disturbances generated by respiration and stomach activity with the crucial positioning of the electrodes due to fetal movements account for the second source of interference. Non-biological interference sources, such as power-line coupling and thermal noise corrupt the recordings as well [1].

Therefore, it is pertinent to devise ways to extract and enhance FECG with the elimination of Maternal ECG component.

II. REVIEW OF THE STATE OF ART

At present various signal processing based techniques are proposed and utilized for FECG extraction and optimal enhancement, such as adaptive filters, wavelet transform, neural networks, adaptive neuron-fuzzy inference, polynomial networks, independent component analysis (ICA), and blind source separation (BSS). ICA and some of methods are recognized a powerful technique for the fetal ECG signal extraction. However, they need multi-channel ECG signals and have structural complexity [2].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

The main thrust of our research is that what characteristics LMS and RLS algorithms during adaptive filtration do exhibit that determine their choice in the implementation of a medical information module?

IV. PROBLEM SOLUTION

Based upon the background study and considering the sensitive nature of the problem we assumed adaptive filtering, the best solution. The prime reasons being, they are adjustable according to finite-time average signal characteristics and a change according to the purpose can be made to achieve the optimal performance of the system.

In the Figure 1 we have some reference signal which can be taken as desired signal \( d(n) \) and input signal \( x(n) \) is being applied to the adaptive FIR filter. The error is calculated as the difference between \( d(n) \) and \( y(n) \). On the basis of calculated error, parameters in FIR filter are changed.

As per our research question additional noise components due to muscular movement of mother internal system as well as fetus can vary during measurement of abdominal signal. Moreover, the strength of fetal heart beat signal is comparatively weaker than maternal one. (Maternal heart beat rate has amplitude 2-10 times greater than that of the fetus.)

For our experiment we used a maternal abdominal signal with the sampling: 5 sec, 500Hz.

Figure 1. Structure of an Adaptive system
For $i = 0,1,2,...$
\[ y(n) = w_i^T x(n) \quad (1) \]
\[ e(n) = d(n) - y(n) \quad (2) \]
\[ w_{i+1} = w_n + \mu e(n)x^*(n) \quad (3) \]

**For an FIR LMS Adaptive Filter of nth-Order**

**Variables:**
- $n = \text{Filter order}$
- $\mu = \text{exponential weighting factor}$
- $b = \text{P(0) initialization value}$

**Initialization:**
- $w_0 = 0$
- $P(0) = d^{-1}$

**Calculation:**
- For $i = 1,2,...$ compute
  \[ z(n) = P(n-1)x^*(n) \quad (4) \]
  \[ g(n) = (\mu + x^T(n)z(n))^2z(n) \quad (5) \]
  \[ a(n) = d(n) - w_{i-1}^T x(n) \quad (6) \]
  \[ w_i = w_{i-1} + a(n)g(n) \quad (7) \]
  \[ P(n) = P(n-1) - g(n)z(n)a(n)/\mu \quad (8) \]

Based upon the definitions of algorithms above and our results presented in Figure 2(b) (c), it is clearly evident that RLS has a much higher rate of convergence. But during convergence LMS algorithm shows a wayward pattern while in RLS case we see steady stabilization.

This property behavior is result of the $n$ number of multiplication and addition that the LMS algorithm requires, while the RLS needs $n^2$ number of the both.

**V. CONCLUSION**

On the basis of the observation we conclude that the LMS algorithm has computational efficiency but slower convergence rate while RLS algorithm with more computational requirements is efficient in the achievement of faster stabilization. In a more detailed study with the change of variables values which determine step size changes in the LMS which became identical to those of RLS in tracking the non-stationary processes despite the exponential nature of RLS weights.

Our further aim is to develop a prototype model as a ‘Medical Information module’ which can be part of a Medical Information system. So, keeping in view the low cost efficient computational devices available it is suggested that RLS algorithm is the good choice keeping in view better convergence rate despite having more computational requirements. Choice of algorithm is going to affect the cost effectiveness as well as technical adaptability of system on the consideration of hardware available.

**REFERENCES**


Modified Tag Anti-collision Algorithm Based on Adaptive Binary Splitting in RFID

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Abstract—In a Radio Frequency Identification (RFID) system, a reader reads a tag when it comes in its interrogation region. When multiple tags in the interrogation region try to communicate with the reader at the same time, reader-to-tag collisions occur. This affects the reading capability of the reader. For avoiding such a situation, there must be some anti-collision algorithms. This paper presents a modified version of Adaptive Binary Splitting (ABS) algorithm for resolving collisions. It has an improved performance than Adaptive binary splitting (ABS) Algorithm, reducing the collision problem and increasing the efficiency as well.

Index Terms—Tag anti-collision, RFID, Modified Algorithm.

I. INTRODUCTION

Due to the rapid increase and advancement in technology, the need for automated identification systems is also increasing day-by-day. Radio Frequency Identification (RFID) is one such system on which the research is going these days. There are different types of tags, such as active tags and passive tags, but in this paper we will only focus on passive tags. The reader sends a signal to activate all tags which are in the interrogation region. This signal energizes the tag circuitry and in response the tags send their serial numbers or IDs to the reader. However, there could be situations where multiple tags try to access the reader at the same time and thus cause reader-to-tag collisions. In this paper we propose a new approach which is based on Adaptive Binary Splitting (ABS) algorithm for resolving these collisions. This approach uses the basis of Time Division Multiple Access (TDMA) and allows the tags to resolve this problem in a more efficient way. By adding the choice of the delay timeslots in Fig. 1 when collisions happen, the performance is enhanced as compared to ABS algorithm.

II. REVIEW OF THE STATE OF ART

Different algorithms related to reader-tag collisions have been proposed and the most widely used is the tree based algorithm. This can further be classified as binary tree protocol and query tree protocol, both of which split tags into two subsets. The major difference between binary tree and query tree protocol is that binary tree uses binary numbers, while query tree uses tag IDs for splitting tags. When tags send their IDs to a reader and the reader checks for collisions, it informs all tags whether there is a collision or not. This is called adaptive binary splitting protocol in [1] and [2]. If some collision occurs, then the tags randomly select a binary number between 0 and 1, and split the tags into subgroups. The whole process continues until there are no more tags left for collision.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

The main problem is that when multiple tags try to communicate with the reader at the same time, then reader-tag collision happens. In this paper, we proposed modified ABS algorithm which will reduce the chance of collisions. Our hypothesis is that if the delay time slot is chosen from a wider range such as this array \( \{0, 1, 2, 3\} \) when collision happens, then the chance of collision will decrease. In ABS algorithm they choose only binary bits which are either 0 or 1. But in modified ABS algorithm we choose randomly numbers from \( \{0, 1, 2, 3\} \), which makes it more efficient than ABS algorithm. The new algorithm was implemented by simulating in MATLAB 7.0.
IV. PROBLEM SOLUTION

In this paper we focus on algorithm shown in Fig. 1 which is based on Adaptive Binary Splitting (ABS). The basic idea is that if no tag responds to the reader, then there will be an idle state. If there is one tag responding then it goes to the readable state and if two or more tags try to communicate with the reader at the same time then it goes to the collision state.

In Fig. 1, Allocate_number (A_num) is defined as the timeslot in which the tag wants to communicate with the reader. Progress_number (P_number) is defined as the timeslot of the system. Terminate_number (T_num) is defined as the end of the timeslot of the system. When P_num is more than T_num, the cycle of detection is over.

In idle state, it means no tag wants to communicate with reader in this timeslot. In this case, the P_num stays constant and then both A_num and T_num of all tags subtract 1.

In readable state, the communication between one tag and the reader is successful. So the P_num moves forward by adding 1.

In collision state, we made some amendments to improve its performance by adding some delay slots. When a collision happens, for all collision tags, the delay slot of each tag will choose one number from the array {0, 1, 2, and 3}. We let each number of the array be chosen randomly by the tag to make sure that 0 will be chosen by at least one collision tag in one time slot to improve the efficiency. Then let the other tags in the region delay the largest number in the array to avoid collisions between the colliding tags and other tags. For the array {0, 1, 2, and 3}, we choose 3 as the largest number in this array. And at the same time, the range of the array for delaying depends on how many tags are within the range of the reader. It can be chosen in different way to deal with different situation. In this paper, we take the array {0, 1, 2, and 3} for example.

Fig. 2 and Fig. 3 are the results of simulation.

From Fig. 2, we can see the performance of modified ABS which is much better as compared to ABS. The number of tags to number of collisions ratio is about 20 percent in the modified ABS algorithm and about 50 percent in the ABS algorithm.

Fig. 3 Relationship between number of tags and number of time slots

From Fig. 3, we can notice that the sum of timeslots is proportional to the number of the tags by applying both methods. And also the sum of the timeslots by using the modified ABS algorithm is nearly the same with the result by using ABS.

V. CONCLUSION AND FUTURE WORK

In this paper, we introduced modified ABS algorithm for passive tag collision problem in RFID system. In our simulation, the result shows that the proposed Modified ABS algorithm reduces the number of collisions and the whole timeslots will not be added compared to ABS. We show that Modified ABS algorithm is more efficient than traditional ABS algorithm.

In proposed Modified ABS algorithm, we only use delay time chosen from a very small array {0, 1, 2, and 3}. In future work, we can extend the range to test the performance of RFID and also test it in the situation that different times are allocated for each idle slot, readable slot, and collision slot in the simulation.

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REFERENCES


Abstract—The emerging growth of computer networks, networked applications and internet has made the communication robust than ever. Network performance need to be addressed by the users, service providers and also the researchers. To assess network performance of the networks many approaches are proposed to solve this issue. This paper introduces a system to measure delay of any two points of monitored network. To assess the design of the system, we implement a prototype of it and evaluate it in three ways: verification of delay, load and performance test and verifies network performance of a network.

Index Terms—Network performance, delay.

I. INTRODUCTION

THE usage of the internet, networked applications e.g. live football match on internet TV, online banking, online shopping etc and distributed applications are growing very fast. New approaches and algorithms are needed to introduce to ensure best quality of the usage of above service. To ensure this best service, network performance has great impact on the quality of service (QoS) and the quality of experience (QoE). Undoubtedly, delay plays one of the major roles to the network performance. In this paper, a system is designed to measure delay of any two points of the monitored network.

II. REVIEW OF THE STATE OF ART

Traffic measurements are one of the fundamental issues for the development and optimization of innovative networking technologies. Network measurement infrastructures support various network-wide active and passive measurement data collection and analysis techniques.

In this project, DPMI (Distributed Passive Measurement Infrastructure) is used to analyze the real time traffic. In DPMI, DAG cards are used to capture the packets. After capturing the packets, it stamps time to the packets and extracts information from the packet header [1].

There are some tools which are involved to monitor network performance. Here we enclosed some of them for the comparison. SNMP based applications are mostly used to monitor the entire network, and not the individual service performance. NetFlow [3] is a tool developed for Cisco routers to track flows; a flow is complete service session. As it works on flows, it cannot report on QoS parameters while the session is ongoing.

A paper [2] is reported with online processing function.

They process all the data online which makes the system quite slow than usual.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Network performance is one of the key factors to satisfy QoS and QoE. Network performance, especially delay greatly influences the overall performance of networked applications. Our research aim is to design a system which can measure delay between any two points of the monitored network. Our research findings are to ensure network performance to each user. How to manage real time network traffic? What could be the method to measure delay of the real time data?

The research aim was also to analyze a system behavior for the measurement of point to point delay. To observe the behavior, the system is required to capture real time network data, read the required information, analyze the packet headers, measure the delay for different modules and different packet loads and observe the capacity of the system, upto which load the system responses properly. This paper helps to find out network performance at different points.

IV. PROBLEM SOLUTION

To solve the problem, a system is designed to measure point to point delay in the monitored network. There are three major components in this section: system architecture, system implementation and result.

A. System Architecture

An efficient system is required to handle large amount of data which can be done by minimizing the time to analyze a packet, efficient data store techniques in a data storage and correlate the information to measure the delay. The entire system can be divided into three components: packet processor (PP), delay evaluator (DE) and database (DB).

Packet processor (PP):

Packet processor reads the packet header captured by the measurement points (MP) of DPMI [1]. Packet source, packet destination, measurement point ID, captured time and captured length are extracted by the PP. PP generates packet identification (hash) for each packet. PP sends all the required information to the delay evaluator to measure the delay.

Delay Evaluator (DE):

Delay evaluator receives huge amount of data from different packet processors. To handle huge amount of real data an efficient memory management is designed. When a new packet arrives it checks the source and destination of the packet. If there is no memory allocated for these particular source and destination, the system allocates a separate memory for that. If the memory already exists then it checks
the packet ID or hash value of the existing packets. If the same packet is not available there, it just inserts the information of the packets. If the same packet already exists, then it calculates delay and deletes the oldest packet on the basis of captured time to free the allocated memory. To ensure faster calculation of delay, the system used an algorithm to insert, sort and deletion of the packets. After measuring delay it sends all the information to the database for further analysis.

The schematic diagram of the system is provided in Fig. 1. There are three major components in this system: PP, DE and DB.

![Fig. 1. System Architecture](image)

**Database (DB):**

Database is used to store all the real time measured data for further analysis and output generation. As the system is expecting huge amount of data in a very small period, the system required an efficient database to handle the real time data. More than one table is maintained to write the data into the database and one particular table is designed to know which table is currently in active mode.

**B. System implementation**

A prototype is tested to observe the proposed design. A software built in C/C++ is used to measure delay between three different points with any number of the user is attached in the measurement points.

**C. Result**

Traces from different sources are used to observe the network performance. Internet data is taken to validate this system. There are two test cases run in the system:

i. Validate the system with different data load on the systems. Here we used 5000 to 35,000 packets to observe the behavior of the system. A trace (provided by BTH) contained 35,000 packet expecting delay for few seconds.

In the fig 2, X axis shows the number of packets and Y axis shows the delay measured by the system. Here we observe that till 15,000 packets, the system work properly. Above 15,000 packets delay is very high.

![Fig. 2. System behavior with different packet loads.](image)

ii. Validate the system with different setup of the system components which have the great influence in the processing time. To increase the process time we use separate three modules for host1, host2 and host3.

![Fig. 3. System behavior with different modules.](image)

In the fig 3, X axis shows the number of packets and Y axis shows the delay measured by the system. In the above graph, it shows that the system works better for 35,000 packets when PP, DE and DB are on the same host1. Above 35,000 packets, the system performs better if the PP, DE and DB are on separate machines.

**V. CONCLUSION**

In this paper, a system is approached to measure delay of packets between any two points of the monitored network. The prototype of the system also validates in different approaches. This paper shows that real time delay of packets in the monitored network can be measured by the proposed system. More scalable memory management can be introduced in future to handle real time data. The analysis of the behavior of this system can also be the future work of this work.

**REFERENCES**


Adaptive Step and Thresholds Algorithm for CDMA Power Control

Bwalya Freelance, Elnourani Mohamed and Moinuddin Syed

Abstract—In this research project we propose a closed loop power control algorithm which establishes quick convergence by the use of both adaptive power correction step and adaptive buffer thresholds. Effective Power control in CDMA systems is crucial owning to the near-far-effect and co-channel interference. The speed with which a power control algorithm converges to an acceptable Signal-to-Interference Ratio (SIR) value defines the quality of service (QoS) and system capacity. The Matlab simulation results show the benefits of the proposed algorithm compared to a classical fixed step with fixed thresholds algorithm.

Index Terms—Adaptive power control algorithm, Code Division Multiple Access (CDMA), log normal shadowing

I. INTRODUCTION

CDMA is a self interference system. This is due to the inherent multiple access coding technique employed. This research project was motivated by the fact that, among the existing algorithms, no optimum solution has been found to effectively manage power control. Thus the aim was to create a suitable algorithm and demonstrate through simulated results, its advantages as regards, convergence, capacity and (QoS).

II. REVIEW OF THE STATE OF ART

Significant research has been done in the area of CDMA power control. Much focus has been on the most effective way to minimize the oscillations of the corrected SIR around the preset threshold [1][2]. Some researchers consider the inclusion of margins around the preset threshold so that if the measured SIR falls within the margins, the transmitter would maintain its transmit power [2][3]. Other researchers consider changing the power correction step adaptively [1][2][3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

While oscillations about the optimum SIR could be minimized, the challenge is; how to ensure that the optimum SIR is reached as fast as possible considering the random nature of the propagation paths for each mobile station. This paper focuses on speeding up the convergence of a forward channel power control algorithm. The research compares the convergence of a classical forward channel power control algorithm that uses fixed step size and fixed SIR buffer thresholds with the proposed algorithm.

The following are the main contributions of the research project:

- Introducing adaptive buffer thresholds to avoid oscillations around the optimum SIR.
- Inclusion of a fast adaptive power correction step size to increase the convergence speed.

IV. PROBLEM SOLUTION

The adopted model uses 100 mobile stations whose positions are random uniformly distributed in a cell of radius 3000 m, log normal shadowing path loss model and a Gaussian random variable with standard deviation of 0.25 dB. The simulations are carried out for both algorithms. The results are then compared to verify the benefits of the proposed algorithm.

A. Fixed Step with Fixed Buffer Thresholds Power Control

The following steps describe the fixed step closed loop power control procedure:

If $SIR_{thl} \leq SIR_{mean} \leq SIR_{thr}$

$Pt = Pt$ "no power up/down"

If $SIR_{mean} > SIR_{thr}$

$Pt = Pt(i-1) - \Delta_{fs}$ "power down"

If $SIR_{mean} < SIR_{thl}$

$Pt = Pt(i-1) - \Delta_{fs}$ "power up"

Where:

- $SIR_{thl}$ is the lower threshold of the buffer
- $SIR_{thr}$ is the upper threshold of the buffer
- $SIR_{mean}$ is the measured SIR at the mobile station
- $P_t$ is the transmitted power
- $\Delta_{fs}$ is the fixed correction step size
- $P_t(i-1)$ is the transmitted power in the previous iteration

The two models use (1) to estimate the path loss using the log normal shadowing model for each mobile station.

$$PL = PL_0 + 10 \times n \times \log\left(\frac{d}{d_0}\right) + X \tag{1}$$

Where:

- $PL$ is the total path loss at distance $d$
- $PL_0$ is the path loss at distance $d_0$
- $n$ is the distance from the transmitter to the receiver
$d_0$ is the reference distance

$X$ is a Gaussian random variable with variance 0.25 dB

The description above shows that the classical algorithm corrects the power by a fixed step each iteration irrespective of how far the measured SIR is from the optimum.

B. Adaptive Step with Adaptive Thresholds Power Control

This method corrects the power adaptively using the previous states of the measured SIR and buffer thresholds. For two successive measured SIR which are higher or lower than the upper or lower buffer thresholds, the correction step is multiplied by a factor $\mu$. If not, the correction step is divided by the same factor. The following description shows the procedure to change the buffer thresholds adaptively.

If $SIR_{\text{old}} \leq SIR_{\text{opt}} \leq SIR_{\text{max}}$

\[
SIR_{\text{new}} = SIR_{\text{opt}} + 0.9 \Delta_{\text{opt}} (i-1)/2
\]

\[
SIR_{\text{old}} = SIR_{\text{opt}} - 0.9 \Delta_{\text{opt}} (i-1)/2
\]

Where:

- $SIR_{\text{opt}}$ is the optimum SIR value
- $\Delta_{\text{opt}} (i-1)$ is the buffer size for the previous iteration

The other procedures are the same as for the previous method.

C. Simulations

The two algorithms were simulated in Matlab. During simulation the following assumptions were made:

- $PL_0$ was estimated at 55 dB for all mobiles;
- Mobiles were assumed to have random uniformly distributed distances from the base station;
- The initial power for all mobiles is 50% of max.

The graphs in Fig. 1 show the convergence of the two algorithms against the number of iterations.

D. Interpretation of Results

In the convergence simulation, due to the adaptive step, the system adapts the step each iteration and speeds up the convergence. Once inside the buffer region the adaptive buffer further forces the SIR to converge towards the optimum SIR. The buffer upper and lower limits only reset to initial values if the measured SIR for a respective mobile falls out of the buffer region in two successive iterations.

On the other hand, the fixed step and buffer thresholds algorithm always corrects the power by a fixed step each iteration. In this case the convergence takes longer. The SIR also keeps fluctuating when it reaches the buffer region.

In Fig. 2, the graph show the percentage of the mobiles that remain within range for the two algorithms under consideration after 50 iterations.

V. CONCLUSION

In this paper, we considered two power control algorithms and investigated their respective convergences and how they affect the QoS. The simulation results in Fig. 1 and Fig. 2 show that the classical algorithm takes longer to converge.

REFERENCES


Abstract—This paper explores the problems occurred during the multicast transmission which lies between unicast and broadcast in an Internal Protocol Suite. The necessity of this study is to compensate for the packet burst or packet out of order problems during multicast transmission. The methods used for solution include SPF (Shortest Path First Algorithm), IGMP (Internet Group Management Protocol), OSPF (Open Shortest Path First) on networks. The issues also included in this research are buffering strategies, available bandwidth, and time stamping methods over the period of transmission.

Keywords—Buffers, Hybrid, Multicast, Topology.

I. INTRODUCTION

MULTICASTING is a way of transmitting to multiple stations on a network regarding an important information by a single send multiple receive operation. As the multicast doesn’t support all the participants to transmit through the internet due to their limited address space restriction in internet, motivated this project to recommend the best methods of transmission which boosts multicast IP datagram delivery.

In this paper, different methods are implemented to prevent packet out of order problems during multicasting. Each method which shows a unique capability in solving a particular problem. Sometimes N number of available methods may give a better solution, but the task of mixture of methods is very crucial during the multicast transmission as explained by different approaches.

II. REVIEW OF THE STATE OF ART

It is important to save the data. A packet travel from sender to receiver through the network, while traveling some packet have been lost. The packet disorder is caused due to low bandwidth. In this paper, the two protocol IGMP and OSPF are studied. There are sort of technique to minimize the packet disorder. But it needs to find out an efficient technique [1].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

SPF algorithm uses a shortest available path to reach a destination before a packet disorder can occur. Here SPF algorithm gives partial solution only, so the addition of remaining process during transmission also important, to get accurate IP datagram delivery. The processes that have been taken care of during the transmission in this paper are:

- Buffering strategies
- Nodes in participation
- Available bandwidth
- Expected time of transmission
- The routing protocols implemented during the design of a network and their scalability.

III. PROBLEM SOLUTION

The available solutions are unique in their nature but these solutions are not able to provide complete solution to packet disorder problems so paper has recommended four methods which can be implemented according to the requirements of a particular network.

A. The first solution is implementing Gigabit Network Ring Topology

The Gigabit Networks technology advises to adopt high bandwidth networks which automatically solve many problems in a structured manner. Here the problem arises when the congestion concept comes out. Normally the top down approach is easy to implement whereas the bottom up approach is difficult and sometimes it is not applicable in particular networks. Why the word not applicable is dominating is that, small companies cannot invest high amount of money on bandwidth requirements and redesign their networks. So this method can be applicable to a network of X to Y stations e.g. 10 to 20 stations.

B. The second solution is to implement SPF algorithm

In shortest path first algorithm link state protocol make use of this algorithm to obtain best paths throughout the network. This algorithm builds the path in such a way to reach all nodes in a network by taking one as root node. This algorithm runs with a frequency less than 5 seconds but the full run occurs only when there is a change in network topology. Here most of the time partial route calculation only happens to make short path updates. This algorithm takes into consideration many points into consideration like 1.Number of nodes 2.number of link in participation 3.degree of nodes in a network 4.Router capability. The SPF algorithm equations which always takes care of shortest path by not considering the repeated nodes into account. The same path is repeatedly...
calculated for best available metric to transfer data in the given network.

Shortest path from node source node to other node
Q = node set of graph
L (i, j) = cost for each node (i,j) (>=0)
Set Q = the source node
P = the no of participating so far
C (N) = cost of path from s to n

- P = \{ s \}
- for each n in Q - \{ s \}
  1. C(n) = l(s,n) (0 if s not directly connected to n)

- while (Q =/= P)
  2. P = P union \{ w \} such that C(w) is the
  3. Minimum for all for all w (Q - P)
  4. -for each n in (Q - P)
     C(n) = MIN (C(n), C(w) + l(w,n))

According to Fig.1, the shortest path to reach from router R1 to R5 is R1->R2->R3->R5-> NOT R1-R2-R3-R4-R5 because the path R1R2R3R5 have cost of 14 whereas the path R1R2R3R4R5 has path of 22. So the metric 14 is preferred instead of 22 in route consideration.

C. Buffer Reservations
The third solution is a method of keeping an amount of memory in advance to keep up fluctuation in data transfer. This method can avoid overflows caused by packet delivery during transmission times. This method requires high amounts of memory.

D. Implementation of OSPF, IGMP in a given Networks.
The fourth solution provides a loop free topology and triggered link state advertisements in a network through OSPF configuration. Here the main advantages of classless protocol and hierarchical design with VLSM route summarization are possible by OSPF configuration in routed network.

As shown in Fig.2, the following network is configured with OSPF routing protocol to solve the multicasting problem either by employing autonomous system or areas. OSPF maintains the link state information of all neighboring routers and their routing table information in a link state database. The typical configuration commands of OSPF are given below OSPF can be employed in large networks through mixed router environment.

According to Fig.1, the shortest path to reach from router R1 to R5 is R1->R2->R3->R5-> NOT R1-R2-R3-R4-R5 because the path R1R2R3R5 have cost of 14 whereas the path R1R2R3R4R5 has path of 22. So the metric 14 is preferred instead of 22 in route consideration.

IV. CONCLUSION
The solution exists are many but changes according to the standards and requirements of Present scenarios. So not only the study of mathematical, or simulation or some other model gives a unique solution but also their implementation is necessary to get good health of a networked environment. After many discussions to found out an efficient solution. It is possible to minimize packet loss during the multicast transmission by using high bandwidth.

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A Terrestrial to Satellite Handover Protocol for LEO Satellite/Terrestrial Hybrid Systems

Chamila A. Ariyaratne, Subash Nemani, and Ravichandra Kommalapati

Abstract—This paper proposes a new terrestrial to satellite handover protocol, that can be incorporated into a low earth orbit (LEO) satellite/terrestrial hybrid system, that uses dual mode handsets. The proposed protocol temporarily diverts the terrestrial calls to idling satellite channels without the need of additional satellite resources. This paper shows, through simulation results, that the proposed protocol can be used to considerably reduce the call incompletion probability of the terrestrial network at times of high traffic loads without disturbing regular satellite activities.

Index Terms—Low earth orbit, Satellite communication, Satellite/terrestrial hybrid system

I. INTRODUCTION

In recent years, the telecommunication industry has focused on the idea of integrating low earth orbit (LEO) satellites and terrestrial networks into hybrid networks to achieve better economic benefits [1]. When the mobile handset is in the coverage area of the terrestrial network, it connects through the terrestrial network and when it moves out of the coverage area, it connects through the satellite network [2].

This paper focuses on a terrestrial to satellite handover protocol that uses only the satellite channels that are being reserved by the regular satellite users but currently not in use, in order to divert the congested terrestrial traffic temporarily to the satellite. The concept of reserved satellite channels is discussed in detail in the later sections. It is shown through simulation results that the proposed protocol will increase the service quality in terms of call incompletion probability.

This enables an operator to offer a better quality of service to the existing users or to accommodate more users while maintaining the same level of service quality without additional resources.

II. REVIEW OF THE STATE OF ART

Unlike the geo-stationary satellites, LEO satellites revolve at rapid velocities with respect to the earth. According to orbital dynamics, such velocities are necessary to remain in lower orbits at altitudes less than 1500 km [3]. This rapid motion of LEOs with respect to earth requires the system to handle inter-satellite handovers during the lifetime of a call, since a single satellite will be available for a ground user for a period of few minutes. Previous work has been done on various inter-satellite handover protocols that guarantee the users with different levels of service quality. Guaranteed handover (GH) service suggests that the user locks or reserves a channel in the next satellite at the time of entering or beginning a call in the current satellite [4]. This guarantees that the user will never be run out of a satellite channel when the handover to the next satellite takes place. Furthermore in [3], elastic handover scheme has been introduced. In elastic handover, the user sends the channel locking request to the next satellite when the remaining time in the current satellite reaches a certain value $T_c$, where its value depends on the quality of service expected by the users. This results in some idling satellite channels which are being locked by the users who are currently in the footprint of the previous satellite.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

As mentioned in the previous section, the satellite handover schemes give rise to some temporary idling channels. This paper examines whether it is possible to use such idling channels in order to divert the terrestrial traffic temporarily to the satellites at times of congestion. Hence it,

- proposes a new terrestrial to satellite handover protocol,
- simulates the proposed protocol on MATLAB,
- presents simulation results which quantitatively show that the proposed protocol can reduce the call incompletion probability of the terrestrial network.

IV. PROBLEM SOLUTION

A. Proposed handover scheme

The details of the proposed handover protocol are given in the flow diagram in Fig. 1. The acronyms used in the flow diagram are defined as: TU – Terrestrial user, TC – Terrestrial cell, SCh – Satellite channel, SU – Satellite user. The protocol flow is same for both new calls and handover calls, hence shown in the same flow diagram.

B. Simulation model

The proposed handover protocol was simulated on MATLAB 7.0.1.2 to evaluate the performance. The parameter used to measure the performance was call incompletion probability of the terrestrial network before and after the proposed handover protocol is implemented. The call incompletion probability is defined as the probability that a call originated in the system is terminated prematurely, either due to being blocked when the call is started or terminated due

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to a subsequent handover failure.

Fig. 1. Flow diagram showing the steps of the proposed handover protocol

The simulation model assumes a strip on earth of 1000 km in width and 4000 km in length which is covered by a very generic and simplified terrestrial cellular network and four LEO satellites. The LEO satellites assume elastic handover scheme for inter-satellite handovers. Both satellite and cellular networks assume fixed channel allocation with 10 channels per terrestrial cell and 6250 channels per satellite footprint. For simplicity, terrestrial cells were approximated by squares of 40x40 km and satellite footprints were approximated by squares of 1000x1000 km. The call duration assumed an exponential distribution with mean call holding time of 180 seconds and new call arrivals were Poisson distributed with mean \( \lambda \) which was varied between 0.5 and 4 calls/min for each terrestrial cell. New call arrival rate for a satellite footprint was kept constant at 625 calls/min/footprint area. The user mobility was modeled by a uniform distribution between 0 to 120 km/h with direction of motion selected randomly to be north, east, south or west.

C. Simulation Results and Discussion

The graph in Fig. 2 shows the call incompletion probabilities for the terrestrial network yielded from simulation for different values of new call arrival rates. The results are shown for the cases before and after the proposed handover protocol has been implemented. The probabilities for the case before the protocol is implemented are obtained in the absence of the protocol, where a new terrestrial call or a handover would simply be failed at times of congestion in the terrestrial network. As the graph shows, at low traffic loads the improvement in call incompletion probability achieved is very small. This is because at such low traffic levels, new call blockings and handover failures are almost zero. But, as the traffic load increases, the improvement achieved becomes more apparent due to the increased number of new calls and handover calls. These calls, that could have failed, are now temporarily diverted to satellite channels.

Fig. 2. Call incompletion probability for the terrestrial network before and after the proposed handover protocol is implemented.

V. CONCLUSION

This paper proposed a new terrestrial to satellite handover scheme, which can be used in a LEO satellite/terrestrial hybrid network. This was shown to improve the call incompletion probability in the terrestrial network, without affecting the regular satellite services or requiring extra satellite resources.

The proposed protocol can be used by operators to offer improved service for the existing customers or to increase the capacity of the network with the available resources while keeping the same level of service quality.

Suggested future work would be to derive a statistical model with closed form equations to predict the probabilities of concern. Work should also be done to assess the feasibility of handling the required network signalling with the processing power of current handsets.

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The authors would like to thank Dr. Wlodek Kulesza for his helpful discussion and guidance.

REFERENCES


The Translation of Braille Written Language to Regular Text using Neural Network.

V. Sai Krishna Dinesh, K. Iswarya Krishna and O. Yeonhee

Abstract— We are proposing a new system which allows the conversion of Braille to regular text using Neural Networks. This paper specially aims for those who want to read Braille written language without knowing Braille. Mobile cameras and scanners are used to take a photocopy of the Braille script. The framework is like recognizing the Braille which is taken by the camera, irrespective of the distances from lens to the script, separation of each character into equal size and finding the desired characters in regular text, using Hebbian-Postulate and Kohonen Network. In future research the Extension of our approach to an audio signal can help the blind.

Keywords— Character recognition, handicapped aids, image texture analysis, neural network applications.

I. INTRODUCTION

Braille is the script which helps the blind to communicate with the world. However, it is not easy for those, who have the blind as a family member, or who want to be assistants to help the blind, to learn and read the Braille. Our paper suggests an approach to help them by converting of Braille written language to regular text using neural networks.

This approach of developing text from Braille can help the blind to live their life a bit independently, if an audio signal is generated from our result. Especially for those who have lost vision when in adult, it is really very difficult to learn Braille.

II. REVIEW OF THE STATE OF ART

There are researchers who used Java programming for obtaining the dots from the Braille image and translating it into text or audio [1]. Character recognition using Neural Network [2] is used for recognizing alphabets from bad or not clear hand writing. Kohonen Network was suggested as one way to cluster inputs into several groups [3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

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Fig. 1. The examples of Latin Braille, such as a, b, c, and d.

The image of Braille character could be recognized and converted to the regular text by using Neural Network.

After taking a photo of the Braille character, specific algorithm or treatment is needed to distinguish Braille script from around noises. The reason why algorithm is needed is that one character of the Braille consists of six dots in one Braille cell. The dots and noise look similar.

The implementation will help the blind to get information through Braille. The system of neural network will reduce the complexity of procedure to convert Braille to regular text.

IV. PROBLEM SOLUTION

Implemented system recognizes the photo image of Braille characters and converts it to English word. The blind take the photo image, when they sense it. Thus, the size of input varies depending on the distance among the lens and Braille. The photo image is assumed as JPEG (Joint Photographic Experts Group). After mentioning Latin Braille characters briefly, we will state the procedure of implementation.

A. Latin Braille

Latin Braille is a written language for the blind to write and read other’s thoughts or information by sensing it through fingertips. One character of Braille is represented as one or several dots of convex and concave format in one cell, three rows and two columns grid. Fig. 1 shows the examples of Latin Braille.
B. Implementation

The translation procedure consists of three steps. During the first step, the format and size of Braille image, taken by a camera, is changed to suit for Neural Network in Matlab. Second step is mapping the distorted onto the desired (clean) Braille script by using Hebbian model, which is the last step to convert the clean script to regular text.

The change of the format and size of input image

As an input of the system, the RGB (Red, Green, and Blue) colors in JPEG image of input are converted as one-dimensional color. Fig. 2(a) shows original image in JPEG format took by a camera. Fig. 2(b) shows the image after converting RGB to one-dimensional of input image.

The size of each character is fixed as 13×12 matrix form. To get 13×12 size of each character, the length of rows and columns are considered.

In the row case, at first, we check the indices of first and last row in image. Then the distance between them are calculated. If the difference is larger than 13, the number of desired row, then the last row of image is subtracted. While the difference is smaller than 13, the row matrix which consists of 1×12 zeros, is added. This whole steps are repeated, until the difference become 13.

In the column case, we need to consider three different cases. Primarily, the distance among the starting point of matrix and first dot is kept two. Secondly, if the distance between the first and second dot’s position is 7, then the whole dots in the Braille cell is positioned only in the left side and not the right side. Last to fix the size of the distance among first dot and third dot is kept two in the Braille cell.

Fig. 2(c) shows the fixed size of input matrix.

Character Recognition with Hebbian Model

For character recognition, we construct a single layer association network with Hebbian Model. The input size of network are 156 (13 X 12) bits. The weight matrices are created by multiplying transpose of it and desired Braille characters, according to (1). Especially, there is no update for weight matrices.

\[ W = XX^T \]  

Where \( W \) is the weight matrix for the system, \( X \) is the matrix of desired Braille, and \( X^T \) is the transpose matrix of desired Braille.

Fig. 3 shows desired Braille as an output from the association network. Comparing Fig. 4 with Fig. 2(c), we can observe that the association network removes the noise and recovers it.

Converting clean Braille to regular text

To convert clean Braille to regular text, we use the idea of Kohonen Network and clustering. The weight matrix is the desired Braille script. After checking the distance between an input vector (156 bits) and each weight matrix, the argument (index or winning unit), which has the minimum distance, is chosen based on (2).

\[ k = \text{argmin} \| W_j - X \| \]  

Where \( X \) is an input vector, \( W \) is the weight matrix for the system, and \( k \) is index of winning unit which has the minimum distance. Operator || \( \| \) \( || \) calculates the norm of the vector.

According to index information, the system can recognize which Braille character belongs to certain Latin letter and display it, which is stored in memory. Fig. 4 shows the regular text that is converted from Fig. 3.

V. CONCLUSION

This paper proposes the system of Neural Network which acts as a tool in getting information from Braille without knowledge of Braille.

The system to recognize Latin Braille is carried out with Neural Network by using Hebbian model and Kohonen Network in Matlab.

The carried out system with Neural Network in Matlab does not need certain algorithm to reduce noise, because the association network gives out closest script to the distorted images.

In further research the system will be tested with more real image which is taken by a camera directly from the real situation and an audio signal can be generated from the text obtained.

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We thank Professor Wlodek J. Kulesza for his constant support and guidance through the preparation of this paper. We extend our thanks to Dr. Nedelko Grbic for his support in the implementation part. We thank our friends who supported us and we dedicate this paper to “The Society for the Blind”.

REFERENCES

Analysis of BER by Improving the Orthogonality of the Channel Coding in OFDM-CDMA System

R. Hussain, A. A. Mamun, P. Prasad

Abstract—In this paper there is an attempt to improve the bit error rate (BER) by improving the orthogonality of the channel code for OFDM-CDMA system. We work on two channel coding schemes for an OFDM-CDMA system, orthogonal and super orthogonal convolutional coding schemes. Super orthogonal coding scheme is the extension of orthogonal coding scheme. We compare both of them and find out which gives less BER at same code rate and same bandwidth B. We also workout the relation between the constraint length K and the BER. We have found that the BER is inversely dependent on the values of constraint length K.

Index Terms—OFDM-CDMA, Orthogonal conv. Coding, Super Orthogonal conv. Coding, BER.

I. INTRODUCTION

The Orthogonal Frequency Division Multiplexing (OFDM) is a multicarrier transmission technique, which divides the available spectrum into many carriers, each one being modulated by a low rate data stream. In OFDM the multiple user access is achieved by subdividing the available bandwidth into multiple channels, which are then allocated to users. However, OFDM uses the spectrum technique more efficiently by spacing the channels much closer together. This is achieved by making all the carriers orthogonal to one another, preventing interference between the closely spaced carriers. Code Division Multiple Access (CDMA) is a spread spectrum technique that uses neither frequency channels nor time slots. With CDMA, the narrow band message (typically digitized voice data) is multiplied by a large bandwidth signal that is a pseudo random noise code (PN sequence). All users in a CDMA system use the same frequency band and transmit simultaneously. The transmitted signal is recovered by correlating the received signal with the PN code used by the transmitter (To combat fading, OFDM can be combined with direct sequence spread spectrum, where the signal is spread over several carriers to achieve frequency diversity R.A. Stirling). The motivation behind this paper is to improve the performance of OFDM-CDMA system by improving the orthogonality of the channel codes. The comparison is taken out at same code rate and same bandwidth.

II. REVIEW OF THE STATE OF ART

After the publication of new decoding algorithm for convolutional codes [2], A. Viterbi proposed a class of orthogonal convolution codes where orthogonal and super orthogonal convolutional coding techniques are the main stream lines. These techniques are compared against different code rates by taking the punctured convolutional code as reference. The comparison is made of the spectral efficiency of the different coding schemes, the bandwidth required and the receiver complexity tradeoffs [1]. It is well known that the orthogonal convolution codes for which a bound on error performance was easily derivable [2].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

The coding schemes we shall consider in OFDM-CDMA systems are orthogonal and super orthogonal channel coding which have a low bandwidth expansion, and so a high spectral efficiency can be achieved with a fully loaded system. In this paper we workout on improvement of Bit Error Rate (BER) in OFDM-CDMA system, the idea is that we keep the code rate and bandwidth in both coding schemes same. Then we analyze that which coding scheme gives the better performance in the sense of BER at the same code rate (R). Also we check the impact of constraint length K on BER. Observations are taken out in additive white Gaussian Noise (AWGN) [1].

IV. PROBLEM SOLUTION

This paper considers two channel coding schemes for OFDM-CDMA system. In this paper no detailed description about the OFDM-CDMA system is discussed.

A. Channel Coding

In orthogonal channel coding [2] the relation between code rate R and the constraint length K is \( R = \frac{1}{2^K} \), where we take \( K = 5 \) and hence the code rate will become \( R = \frac{1}{32} \). Super orthogonal channel coding [2] is the extended version of orthogonal channel coding in which we take \( K = 7 \) also the relation between the code rate and constraint length is \( R = \frac{1}{2^{K-2}} \) [1], hence the code rate is same for both schemes.

B. Code Evaluation for BER

In this section we evaluate the codes for BER, considering orthogonal channel code. The BER is obtained by differentiating the code transfer function \( T(W, B) \) [1][2].

\[
BER \leq \frac{1}{2} \frac{dT(W,B)}{dB}
\]  (1)
Where \( r = \frac{1}{R} \), relative bandwidth \( B = \frac{1}{W} \), \( W = Z T_r \), \( Z = e^{\left( -\frac{E_b}{T_0} r \right)} \)

For the orthogonal coder the derivative of the code transfer function is given by

\[
\frac{dT(W,B)}{dB} = \frac{W^k (1-W)^2}{(1-2W+W^2)^2}
\]  

(2)

And for the super orthogonal channel coding,

\[
\frac{dT(W,B)}{dB} = \frac{W^{k+2} (1-W)^2}{(1-2W)^2 (1-W^2)^2}
\]  

(3)

C. System Parameters and Simulation Results

The OFDM-CDMA transmitter uses a 512 point IFFT where we have 8 sub-systems using 32 length Walsh codes. For an encoded data rate of 16 kbits/s the data rate per carrier is 2 kb/s with a transmit bandwidth of 1.025 MHz. We assume there is sufficient dispersion in the channel so that the interleaver can achieve perfect interleaving. This is achievable in our system if the channel dispersion is greater than 14 \( \mu s \) [1]. When comparing different coding methods the uncoded data rate is fixed at 16 kb/s and thus the different coding schemes occupy different bandwidths. We could compare system with the same transmission bandwidth, but this would require different uncoded data rates or different levels of processing gain. Making a comparison with different uncoded date rates is difficult due to the different performance of the required speech codes. Making a comparison with different processing gains in OFDM-CDMA is difficult due to the different levels of diversity [1].

Fig. 1 shows that the simulation results of orthogonal and super orthogonal channel coding at \( K=5 \) and \( K=7 \) consecutively at the same code rate, if we compare these two results then we will find that the super orthogonal schemes give better BER performance then orthogonal scheme.

![Comparison of both tech with same data rate](image1)

In Fig. 2 if we assume that our system is leaner and considers this is an ideal case then we observe that the value of BER is exponentially decreases with increase in \( K \). Hence at large PN (chip sequence) sequence we can increase our users and provide service with more accuracy and less BER, but our system becomes more complex. Like to produce a long PN sequence we need complex equipment. This result may arise many questions like how is it possible that if we increase the length of code results decrease in BER at the receiver end in a noisy channel. This result may question mark the accuracy of equations presented by A. Viterbi about the orthogonality of the proposed schemes.

V. CONCLUSION

We can improve our system performance by improving the orthogonality of the channel coding schemes, coding put a deep impact on the system performance especially when we talk about the spread spectrum then orthogonality and the improvement of orthogonality will become a core issue. By improving the orthogonality of the generated codes we can improve the BER. In its present form, super orthogonal coding is a good choice in a channel with a low \( E_b/N_0 \). Dependency of BER on the value of \( K \) is also an interesting result.

REFERENCES


Abstract—In this paper we study the network mobility solutions and the implementation of a mobile network in a real life scenario. First we introduce the mobile network requirements and scenarios, and then we study the Mobile Internet Protocols and its modification to support network mobility. We implement our solution in a moving train, and then conclude with some hints to improve the mobile network performance.

Index Terms—Mobile IP, Network Mobility (NEMO), Mobile Router.

I. INTRODUCTION

The emergence of mobile networks get more critical in today’s fast world, a set of hosts move together as a unit such as on trains, ships and aircrafts, Persons own more than one mobile device such as a pocket PC, a cell phone and laptop; these devices are moving with the user form a small scale mobile network called a Personal Area Network (PAN). An example of a larger scale mobile network is the access networks deployed on moving vehicles such as trains, buses, airplanes and ships. The protocols for mobility support should be extended from supporting a single mobile unit to support an entire mobile network [1].

Mobile IP protocol (MIP) is the mobility solution on the global Internet that allows single nodes to maintain ongoing communications while changing several networks [2]. The network mobility support provided by MIP and MIPv6 is not sufficient as some devices in a mobile network such as sensor cannot run these complex protocols, and the device once it has attached itself to a Mobile Router (MR) on a mobile network may not see any change even if the network becomes mobile. The solution can be obtained by modifying the current MIP to support the movement of a network.

II. REVIEW OF THE STATE OF ART

The objective of the IETF-NEMO (Internet Engineering Task Force) working group is to develop mechanisms that would provide permanent internet connectivity to all the mobile networks via their permanent IP addresses. IETF-NEMO is developing a basic protocol that ensures continuous connectivity to the mobile networks without considering route optimization. This protocol runs on the mobile router and its home agent and uses the same mechanisms as the host mobility protocols [1]. The Overdrive European Project has designed and developed IPv6-based vehicular (mobile) network to provide Passengers with internet access via the MR installed in the car, which is capable of moving between 3G, WLAN, and DVB-T access networks in an IPv6-based B3G system [3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

We mean by network mobility the movement support of IP sub networks, giving it the ability to change its point of attachment to the internet (change the IP topological location). The main questions arise as how to handle the mobility of the entire network consisting of several mobile routers and attached hosts moves as a single unit and still keep the ongoing connection. Then how to deal with the different scenarios involved during the network mobility. Our main contribution is how to implement it in a real life travelling train with passengers using their laptops to connect to WLAN access points installed throughout the train using a mobile router to provide connectivity to the internet.

IV. PROBLEM SOLUTION

The basic approach to network mobility support is that a mobile router acts on behalf of the nodes that are present within its mobile network. A network changes its attachment point from home network to another network using the procedure:

1. One router within the mobile network indicates to its Home Agent (HA) that it is acting as a MR.
2. MR informs the Home Agent (HA) of the new mobile network addresses (Network Prefix and care of address); these new addresses allocated are used by Home Agent to intercept the packets addressed to the MR and tunnel them to the mobile network at its care of address.
3. HA uses encapsulation and tunneling to forward the packet from the home network to the mobile network.
4. MR decapsulates the packets and then forwards them to the mobile node within its mobile network.
5. Packets in the reverse direction (MR to HA) are also tunneled via MR to HA where the packets are decapsulated and then forwarded to the CN (correspondent node).

Some improvements of binding cache, advertisements and routing table’s management are desired in both MR and it’s HA.
Figure 1: Packet delivery in the train scenario

A. Travelling Train

To illustrate the packet delivery in case of train’s mobile network consider a correspondent node (CN) wants to communicate with a mobile node in the passenger’s wireless LAN travelling within the train. This scenario is shown in figure [1]. First MR2 (Passenger’s WLAN) has registered the care of address obtained from MR1 (Train’s LAN) with its HA-MR2 (home LAN). Furthermore MR1 has registered the Passenger’s WLAN address and it’s care of address with its HA-MR1 (Main Station LAN). A packet from correspondent node is initially sent to the home address of the mobile node (home LAN). In the home network the packet is intercepted by the HA-MR2 which indicates that the network prefix for the mobile node is at the train’s LAN care of address. The packet is then encapsulated by HA-MR2 and is tunnelled to the train’s LAN care of address (Main Station LAN). At the train’s LAN home network the packet is intercepted by the HA-MR1 which determines that the packet again needs to be encapsulated and tunnelled to the care of address of the passenger’s WLAN. When the encapsulated packet reaches MR1, the outer care of address is recognized as its own prefix and then MR1 decapsulates the packet and forwards it to MR2 which then again decapsulates the packet and determines to which node the packet belongs to. Then finally forwards the original packet to the correct mobile node in the passenger’s WLAN. Packet from the mobile node back to the CN travels the same path in the reverse direction.

B. Result Validation

We apply this procedure on modified MIP protocol simulation using network simulator NS2.33 as implementation environment to make it more suitable with the train scenario. Figure [2] shows the simulation work space which contains the home agent, the mobile network and the foreign networks. Figure [3] shows the mobile network performance as a measure to the packet losses while the network is moving. Packets can be delivered successfully in both directions with different delays using bi-directional tunneling. The packet losses increased during the network handover from one network to another and this due to the advertisement and new IP address agreement delay.

Figure 2: NS2.33 Work space

Figure 3: Mobile network performance (packet loss vs. time)

V. CONCLUSION

This paper showed the practical realization for network mobility scenario of a moving passenger’s train connecting to the internet using a mobile router. As shown, providing basic network mobility support requires some extension to the MIP operations of the Mobile Router and its home agent. Providing route optimization for nodes within a mobile network is quite challenging, the mobile network within the moving train can still be connected to the home network by using the MR-HA bi-directional tunneling procedure. Reducing the packet losses must be considered in the future work.

The Future of network mobility at IP level can be brighter, more complex scenarios can be studied on a larger mobility scale.

REFERENCES

Forwarding Measured Data Using ZigBee: A Simulation Technique for Wireless Sensor Network Design

Chandan Kumar Khatri, Kamran Yousuf, Ngugi Lawrence Chege

Abstract—ZigBee is an emerging wireless technology that has vast potential for use in Wireless Sensor Networks (WSNs). This paper describes a simulation method to determine appropriate ZigBee network parameters to fit design requirements. The impact of increasing the number of nodes and packet sizes on end-to-end delay and network throughput is investigated.

Index Terms—Self-Configuring Networks, Wireless Sensor Networks (WSNs), ZigBee.

I. INTRODUCTION

ZIGBEE is a simple, reliable wireless network technology that is suitable for control applications. It provides reliable, robust, self-configuring and self-healing networks [1].

ZigBee is preferred in sensor networks for its capacity to accommodate more devices compared to other technologies such as Bluetooth.

Two specifications of the ZigBee protocol stack exist. In the IEEE specification, only the PHY and MAC layers are defined while the ZigBee Alliance specification adds Network Layer (NWK) and Application Layer (APL) to the stack profile defined by the IEEE [4 Fig-2].

ZigBee devices can be defined as Reduced Function Devices (RFDs) or Full Function Devices (FFDs). FFDs can communicate with any other device assuming the role of coordinator or Router. RFDs can only communicate with FFDs. FFDs are, therefore, more complex in their design and consume more power.

II. REVIEW OF THE STATE OF ART

The transmission range of most ZigBee devices is the range of ~100m (outdoors) which is inadequate for deployment in vast industries. Multi-hop technology can increase network coverage to large geographical areas [2]. This, however, increases network latency.

ZigBee has been touted as having greatest potential for use in class 4 and class 5 applications [3]. But routing mechanisms for ZigBee network formation, addressing of nodes and performance in delay sensitive applications are yet to be adequately addressed.

It is primarily because of these concerns about delay that players in industry have been reluctant to adopt ZigBee technology to replace wired protocols such as HART.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

In this paper, we investigate the performance of a meshed ZigBee network modeled in OPNET to forward measurement data from instruments in the plant to a central control unit.

Fig. 1. Meshed Network Model

Specifically, effects of varying the number of nodes and packet size on network performance are examined.

The main contribution of this paper is the development of a method of determining suitable ZigBee network sizes to meet design requirements through the use of OPNET simulation software.

IV. PROBLEM SOLUTION

A ZigBee mesh network was modeled and simulated in OPNET. This network included ZigBee nodes and one ZigBee Coordinator. The ZigBee Coordinator was tied to the Control unit and collected data from all nodes in the network.

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Number of nodes</th>
<th>Packet size (bytes)</th>
</tr>
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<tbody>
<tr>
<td>1</td>
<td>20</td>
<td>256</td>
</tr>
<tr>
<td>2</td>
<td>20</td>
<td>512</td>
</tr>
<tr>
<td>3</td>
<td>40</td>
<td>256</td>
</tr>
<tr>
<td>4</td>
<td>40</td>
<td>512</td>
</tr>
</tbody>
</table>

Four scenarios were formulated in OPNET to simulate different network environments as shown in Table-1. Packet interval time, transmit power, number of
retransmissions, acknowledgement timeouts etc were set to their default values.

The simulation was run over a period of 30 minutes. As a result, the following statistics were collected:

- Packet delivery rate
- End-to-end delay

A. Packet Delivery Rate (Packets sent vs. Packets Received)

As time progressed, more nodes joined the network, each placing its own data into the network. This explains the rapid rate of increase of packets sent. The number of packets sent per second flattens out after all nodes have joined the network.

Simulation results indicate 100% packet delivery in the initial stages of the simulation. Thereafter, a fairly constant rate of packet loss (difference between packets sent and packets received) is observed for all scenarios.

It is worth noting that figures reported for packets received include retransmitted packets. Undelivered packets were only recorded after 5 failed retransmissions.

It was observed that other factors remaining constant; the ZigBee network is able to provide a predictable packet delivery rate regardless of packet size or number of nodes.

B. End-to-end delay of application data

It was observed that end-to-end delay for all Scenarios was greatest at the start of the simulation and thereafter fell to the range of milliseconds as simulation time progressed.

This may be attributed to lack of optimum routing paths for packets in the initial phases of network establishment. During this period, network control and management data as opposed to measurement data accounts for a larger portion of the traffic.

Delay is greatly affected by number of nodes with scenarios 3 and 4 exhibiting the highest end-to-end delay in the initial stages. By comparison, packet size has a less significant effect on end-to-end delay.

V. CONCLUSION

Delays of 3 to 4 seconds are unacceptable in many control applications especially when the control cycle is less than 1 second. For large networks, a hierarchical system could be employed so that smaller ZigBee networks aggregate their data on an intermediate device with higher bandwidth e.g. Wi-Fi for onward transmission to the control unit.

Through the use of simulation tools such as OPNET, an appropriate size for ZigBee networks can be determined depending on the delay requirements of each particular network.

Future work in this area may include incorporation of other wireless technologies to sensor networks with a view to improve network performance.

VI. ACKNOWLEDGEMENT

Members of group Linkoping would like to recognize the invaluable contribution made by Prof. Wlodek Kulesza towards the successful completion of this paper especially for his assistance in securing OPNET in the computer lab.

VII. REFERENCES

Abstract—This paper describes Reduction in Peak-To-Average Power Ratio (PAPR) in the Orthogonal Frequency Division Multiplexing (OFDM) signal by repeated clipping and frequency domain filtering. OFDM is a multicarrier transmission so there is a possibility that signals can get constructive interference and results into high peaks. To reduce these high peaks we perform clipping operation on the oversampled input signal followed by the frequency domain filtering to eliminate out of band signal power which is produced as a result of over sampling and we repeat this process for few iterations until we get signal with low peaks.

Index Terms—OFDM, PAPR, Frequency Domain Filtering, Repeated Clipping and Filtering.

I. INTRODUCTION

OFDM is being widely used in wide band wireless communication like asymmetric digital subscriber line, digital audio video broadcasting and local area network standards because of its efficient use of bandwidth. In addition to this OFDM has high tolerance to multipath signals and also it avoids intersymbol interference (ISI).

But major drawback of OFDM is that it has high peak amplitude signals at some points. OFDM signal consists of lot of Independent modulated subcarriers that leads to the problem of peak to average power ratio (PAPR). If all subcarrier come with same phase, the peak power is N times the average power where N is the total number of symbols in an OFDM signal. So we can not send this high peak amplitude signals to the transmitter without reducing peaks. Because power amplifier used for the transmission has non linear nature which may cause inter modulation and change in signal constellation.

To reduce PAPR we used repeated clipping and filtering operation. Clipping operation is performed to cut the high peaks of the signal and filtering is performed to eliminate the out of band signal power. And finally the results in the graphs show the probability of occurrence of signal corresponding to the power during different iterations.

II. REVIEW OF THE STATE OF ART

An OFDM signal consists of a number of independent modulated subcarriers, which can give a high PAPR when added up coherently. When N signals are added with same phase, they produce a peak power that is N times the average power. A high PAPR bring disadvantages like an increased complexity of the analog to digital and digital to analog converters and reduced efficiency of radio frequency (RF) power amplifier [2].

In a simple way we can reduce PAPR of OFDM signals by clipping the high amplitude peaks. But without performing interpolation before clipping it be causes peak regrowth. To avoid peak regrowth signal should be clipped after interpolation. However this causes significant out of band power. So we use frequency domain filtering to minimize out of band power. Filtering also cause peak regrowth, although this is less then for the case of clipping before interpolation [1].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

As we discuss earlier when we perform clipping operation it significantly increase out of band power in the OFDM signal. For this we check how can we reduce PAPR of the OFDM signal with or without minimum effect on the inband signal power.

IV. PROBLEM SOLUTION

To overcome this problem we used repeated clipping and filtering schema to reduce PAPR. Figure 1 shows how the repeated clipping and filtering is performed. We used serial to parallel converter to convert serial input data having different frequency component which are base band modulated symbols and apply interpolation to these symbols by zero padding in the middle of input data to avoid peak regrowth [1].

These peak regrowths are generated when we perform clipping operation to cut high peak amplitudes. But interpolation results in out of band power, to reduce this out of band signal power we perform frequency domain filtering. The filter consists of two FFT operations. Forward FFT transforms the clipped signal back to discrete frequency domain. The inband discrete components are passed unchanged to inputs of second IFFT while out of band
components are nulled [1] which are represented by 0’s in a clip and filter block as shown in figure 1. The filtering also causes some peak regrowths. However this is much less than for clipping before interpolation.

The whole procedure as shown in figure 1, is for one iteration of clipping and filtering. If we repeat this clipping filtering operation for some iterations then we can significantly reduce PAPR with or without negligible peak regrowths. The dotted line in the Figure 1 shows repetition of clipping and filtering operation.

In our solution we used two modulation schemes for the baseband signal quadrature amplitude modulation (16-QAM) and quadrature phase shift keying (QPSK) to check the impact of these modulation schemes on the PAPR as shown in figure 2 and figure 3.

We used MATLAB as a simulation tool. For the same number of iterations, we use two different modulation schemes as shown in figure. The Figure 2 shows the probability of occurrence of OFDM signal corresponding to the different power level using QPSK modulation scheme. Whereas figure 3 shows the probability of occurrence of OFDM signal corresponding to the different power level using 16-QAM modulation scheme. It is clearly visible from the graphs that as we perform more and more clipping and filtering, the power of the OFDM signals are going to reduce. From both graphs it is clear that the power of the OFDM signal for QPSK is less than as compared to 16-QAM because in 16-QAM we have 16 different power levels to modulate baseband signal whereas in QPSK we just need four different power levels to modulate baseband signal.

V. CONCLUSION

This paper shows how can we significantly reduce PAPR by repeated clipping and filtering technique with or without getting very low amount of out band signal power and peak regrowths. So there is a trade off between costs of the system in terms of number of iterations and PAPR. From the above results we also conclude that its better to use QPSK modulation scheme for baseband signal modulation instead of using 16-QAM modulation or higher modulation schemes where we are concerned in reducing power in an OFDM signal.

REFERENCES


Abstract—Multi-user orthogonal frequency division multiplexing access is a promising technique to achieve high downlink speed in modern wireless and telecommunication system. If the resource allocation of multi-user OFDM is equal to all sub channels, that does not give the guaranty of different levels of service for different demands of each user. Intelligently allocate the resource in according to the channel requirement with some constraints provides that with almost no effect in the efficiency of the channel. In this paper we prove that as it shows the efficiency comparison by applying equal and proportional constraints.

Index Terms—Dynamic resource allocation, Multiuser OFDM, Proportional constraints, Quality of Service.

I. INTRODUCTION

Orthogonal Frequency Division Multiplexing Access (OFDMA), also known as Multiuser-OFDM, is a promising technique for both modulation and multiple access, for the fourth generation of the wireless communication systems. In the OFDM system, there is only one user who can utilize the subchannels (subcarriers) of the OFDM symbol. While in OFDMA, the users share the subcarriers of the same symbol. OFDMA system uses static or dynamic resource allocation to distribute the subchannels among the users. In static resource allocation, the subchannels are assigned to the users manually. On the other hand, the dynamic resource allocation adaptively assigns the subchannels according to the channel condition in order to achieve the highest possible capacity.

In MU-OFDM systems, the multi-carrier transmission methods gives main focus to support reliable and high speed wireless communications. An efficient OFDMA sub-carrier allocation scheme should use spectral as efficiently as possible and achieve minimum cost of service, which meets user quality of service requirements. The allocation technique involves margin and rate adaptive, and this paper focuses on the rate adaptive.

This efficient allocation does not guarantee every user, to get any number of subcarriers. Because it only depends on the user’s channel state and condition. To solve this problem, we can impose additional constraints into the original dynamic allocation algorithm to achieve the fairness on equal basis. These constraints can be referred as equal rate constraints. In addition to achieve fairness, the QoS requires service level differentiation which needs to impose proportional assignment for the subchannels among the users. So instead of using equal rate constraints, the proportional rate constraints can be imposed.

In this paper, we simulate the subcarrier allocation algorithm with equal and proportional rate constraints. We studied the impact of applying the proportional constraints on the channel capacity which reflects on the whole system efficiency.

II. REVIEW OF THE STATE OF ART

OFDM is a robust modulation technique; it is a process of digital modulation that is used in wireless technology today [1]. Multi-user OFDM provide multiple access of OFDM symbol and now-a-days resource allocation is applied in OFDM technique. There are two kinds of resource allocation scheme such as fixed resource allocation [1] and dynamic resource allocation [2]. The fixed resource allocation is based on TDMA and FDMA, it assigns an independent time slot or sub channel for each user. Dynamic resource allocation is based on TDMA and FDMA, it assigns an independent time slot or sub channel for each user. Dynamic resource allocation allocates the subchannels and the power to the users dynamically where we can use optimization techniques to determine the best allocation for these resources.

The optimization techniques can be classified into two kinds such as Margin-adaptive (MA) technique [2] and Rate-adaptive (RA) [3]. MA technique objective is to provide resource allocation with minimum power consumption and RA technique aims to provide the allocation with maximum achievable data rate per user. Researchers proposed two RA optimization problems to solve the channels and power allocation in OFDMA. Firstly, Jang and Lee proposed the rate maximization problem in [3] and used the solution to suggest a practical allocation algorithm. And then, Rhee and Ciffi studied the max/min problem [4] and they applied the solution to provide maximum capacity in MU-OFDM systems. In [5], they proposed an algorithm that achieves fairness in the resource allocation among the users by adding constraints. The kind of constraints they applied is the proportional rate constraints which also can provide different-level of QoS. This paper extends this work by introducing further study on the effect of applying this kind of constraints on the overall capacity of the OFDM channel.
III. RESEARCH QUESTION AND PROBLEM STATEMENT

The frequency spectrum of the OFDM divides all the bandwidth into orthogonal and non-interfering sub-carriers and the data streams are multiplexed on the different sub-carriers. The fixed resource allocation scheme fails to exploit multiuser diversity, results in failing performance of system. So the dynamic resource allocation will be taken into account, to maximize the overall rate to achieve the maximum capacity under the total power constraint. The OFDM multiplexes low rate data sub-streams from the single user onto all sub-carriers, and it can also multiplexes data streams from different users onto subsets of the sub-carriers. So the rate maximization problem under the power constraints will be taken into account.

It is observed that, while assigning the sub-channels to the users, the user will get most of the resources (sub-channels), who have higher average channel gain. And the user with lower average channel gain may get less or no sub-channels. As solution of this fairness issue, the proportional rate constraints technique has been proposed.

It has to be proved, while putting proportional or equal constraints on sub-channels; the efficiency of the system will remain almost the same, while QoS will be affordable with the proportional.

IV. PROBLEM SOLUTION

A simulation has been done using MATLAB to implement the proposed comparison in the model of the OFDMA system. The simulation runs the algorithm for both methods to get the simultaneous capacity of the channel for each number of users. The channel is modeled as a frequency selective multipath channel. It consists of six independent Rayleigh multipaths. Also the channel is assumed to have a delay spread of 5μs and Doppler of 30 Hz as maximum values. To update the channel and power allocation information, the sampling period is chosen to be 0.5 ms. The OFDM symbol is assumed to have the total of 64 subcarriers with 1 W total power. The bandwidth of the channel also assumed to be 1 MHz with 33 dB subchannel SNR, and bit error rate BER \( \leq 10^{-3} \).

The number of users is chosen to increase from 2 to 16 with a step of 2. For each number of users, the simulation runs 1000 channel realizations with 100 time samples for each one.

Figure .1, shows the total capacity of the channel for each number of users in a downlink multiuser OFDM symbol with two curves represent the two kinds of constraints under comparison. The blue-dotted line represents the capacity of the channel when applying the proportional rate constraints and the other represents the capacity of the channel with equal rates constraints. The graph also shows the maximum variation in the capacity between the two kinds does not exceed 1.2 bits/s/Hz and it is almost identical along the whole graph.

V. CONCLUSION

In our paper, while assigning equal and proportional rate constraints to the subcarriers resulting almost no change in efficiency regarding subcarrier capacity. Using proportional rate constraints, it is observed that this method upgrades the fairness requirements of the system as well as different level of service demands by the users. This method also plays an important role in the QoS aspects.

In the same background, researchers can propose other constraints to improve the system usability while observing the impact of those constraints on the overall efficiency of the system as a future work.

REFERENCES

Abstract—Wireless Sensor Networks (WSN), with hierarchical organizations have recently attracted a lot of attention. A critical issue in wireless sensor networks is power saving since the nodes are energized by battery power which are not generally rechargeable or replaceable. Optimization of sensor node lifetime may be achieved by holding control over the transmission and retrieval of data. The data collected by WSN nodes are correlated, utilizing this concept with the principles of entropy compression, we propose a simple algorithm which makes the node to work efficiently.

Index Terms—Wireless Sensor Networks, Data compression.

I. INTRODUCTION

SENSOR nodes are geared up with batteries which cannot be often changed or recharged. Due to this energy become a primary concern in the deployment of Wireless Sensor Networks (WSN’s). Maximum energy in a node is consumed by the radio transceiver. Hence, an approach to reduce power consumption is by controlling the radio system. The radio system in a WSN node consumes maximum energy which the node harvests on, and makes the node inefficient after certain duration of time. Duty cycling scheme defines coordinated sleep/wakeup schedules in the nodes deployed. On the other hand, in-network processing assists in compression of transmitted data [1]. In our paper, we consider a simple compression algorithm which can be embedded on a WSN node which compresses the data transmitted and received by the sensor node. We obtained considerable compression ratios which make the WSN node work efficiently and we compare our results with the results obtained by the standard compression schemes gzip [7] and bzip2 [8].

II. REVIEW OF THE STATE OF ART

We reviewed many methodologies and papers which worked on energy conservation of a WSN node by clustering, data aggregation, packing algorithms like Best Fit, Harmonic algorithms for data packing, efficient routing algorithms and data compression in which we found an interesting concept of conserving the energy by data compression. They use a lossless algorithm for cluster-model WSNs and they discussed about embedding lossless compression algorithms into sensor nodes [2].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

The technique of compressing data will be a worthy aid in power saving only when execution of data compression algorithms will not require energy greater than the energy saved in reducing transmission of data. In wireless battery powered devices, compression of data before transmitting will increase the amount of energy consumed. Compression algorithms help in compressing data, but not in saving energy [3]. In this paper we consider a data compression algorithm which works on data redundancy and data irrelevance, a simple flowchart of the algorithm is showed in Figure 1. The flow chart explains how compressed data is adapted from coding and data reduction. In coding removal of redundancy is done and in data reduction removal of irrelevancy is done.

![Flowchart](image)

Fig. 1. Flowchart

IV. PROBLEM SOLUTION

A. DESIGN OF COMPRESSION ALGORITHM

In a sensor node, each captured measurement $c_i$ acquired by a sensor is converted by an Analog to Digital converter (ADC) to a binary representation $b_i$ on $X$ bits, where $X$ is the resolution of the ADC. For each new capture $c_i$, the compression algorithm computes the difference of the binary representations obtained from the captures $d_i = b_i - b_{i-1}$. The
difference $d_i$ is given as the input to an entropy encoder and in order to compute $d_0$ we assume that $b_1$ is equal to the central value among the 2X possible discrete values [4]. Each difference $d_i$ is represented as a bit sequence $s_i$, composed of two parts $l_i$ and $m_i$, where $l_i$ encodes the number $z_i$ of bits needed to represent $d_i$ and $m_i$ is the representation of $d_i$ when the difference $d_i$ gets to zero then $z$, becomes zero, or else $z = \log_2(|d_i|)$. Thus, at most, $z_i$ is equal to $X$. Code $l_i$ is a variable-length symbol generated from $z_i$ by using Huffman coding. The basic idea of Huffman coding is to map an alphabet to a representation for that alphabet, composed of sequences of bits of variable sizes, so that symbols that occur frequently have a smaller representation than those that occur rarely [5]. In our case, the symbols are $X+1$ and probabilities decrease with the increase of values.

The $p_i$ part of bit sequence $s_i$ is a variable-length integer code generated in a sequential steps followed as, if $d_i > 0$, $p_i$ corresponds to the $z_i$ low-order bits of the two’s complement representation of $d_i$, if $d_i < 0$, $p_i$ corresponds to the $z_i$ low-order bits of the two’s complement representation of $(d_i-1)$, or by if $d_i = 0$, $l_i$ is coded as 00 and $p_i$ is not represented.

The process used to generate $a_i$ guarantees that all possible values have different codes. Once $b_s$ is generated, it is appended to the bit stream which forms the compressed version of the sequence of captured measurements $c_i$.

### B. Validation

There were a certain set of Network simulation software’s where a virtual WSN can be created and the simulated for a data set. As our main aim was to develop an algorithm in an interface which can be embedded on to a WSN node like embedded C. By simulations performed on Avrora [6], an instruction-level sensor network simulator, we have observed that the execution of the algorithm requires considerable number of instructions. By transmitting a single bit, WSN node needs energy which is comparable to the execution of a many instructions. So by just stopping a single bit by compressing original data certainly implies to reduce the power. The performance of a compression algorithm is usually computed by using the compression ratio defined in equation (1). Where comp size and org size are the size of the compressed and the uncompressed bit stream respectively.

$$\text{comp ratio} = 100 \times (1 - (\text{comp size} / \text{org size})) \quad (1)$$

<table>
<thead>
<tr>
<th>TABLE I</th>
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<tbody>
<tr>
<td></td>
</tr>
<tr>
<td>orig size</td>
</tr>
<tr>
<td>comp size</td>
</tr>
<tr>
<td>comp ratio</td>
</tr>
</tbody>
</table>

In our case we assume that data samples have to be transmitted to the data collector are transmitted by using the lowest number of packets in which uncompressed data samples are generally byte aligned, in our dataset, temperature and relative humidity samples are represented by 16-bit unsigned integers. We consider that each packet can contain maximum 25 bytes of payload [5], where the uncompressed versions of temperature and relative humidity data need 116 packets each to be sent from the node to the data collector, but the compressed data sets need less packets to be sent for considerable power saving. For judging the worth of our algorithm we are comparing our results with a well known lossless compressions, in Table II we also show the results obtained by applying two well-known compression algorithms, namely gzip [7] and bzip2 [8], to the same datasets. By comparing these results with Table II, we can observe how our algorithm provides better compression results compared to both gzip and bzip2. Figure 2 shows the comparison of our results with the standard schemes.

### V. Conclusion

In this paper we have used a simple data compression algorithm to minimize the storage and computational resources on a WSN node. We evaluated the algorithm with temperature and relative humidity data and got a compression ratio of 26.78% and 39.82% respectively and then we have compared our results with two previously proposed compression algorithms gzip and bzip2 in WSNs. Hence, we have shown that our algorithm has obtained better compression results then gzip and bzip2.

### Acknowledgment

A good work is always an effort of good thoughts. The authors would like to thank our respectable Prof. Wlodek Kulesza for motivating and supporting us, and our friend Mr. Rakesh Reddy Banda [M.Sc DC, USyd, Australia].

### References


RSVP Extension for Integrated Services Packet Network in UMTS

Yang Wen Wen, Nadeem Ahmad, Muhammad Awais

Abstract—The task at hand is to add an extension of resource reservation protocol (RSVP) to keep Quality of Services (QoS) architecture for a Universal Mobile Telecommunication System (UMTS) and class based QoS traditional methodology. We then identify the existing resource reservation protocol in a mobile network and it illustrates integrated services model offering end-to-end QoS guarantees in UMTS network on the basis of extension solution in RSVP for a numerous types of real time applications.

Index Terms—RSVP, QoS, UMTS

I. INTRODUCTION

The aim of the 3G network is to present a wide range and diversity of services including the multimedia services and internet facility to the mobile users.

The cellular network is hurriedly taken on suitable network model for supporting packet data service. A key component of this packet data services model is QoS. There are four QoS classes: conversational class, streaming class, interactive class and background class.

The major QoS Provisioning mechanisms are multi protocol label switching (MPLS), differentiated services and integrated services, which are proposed to guarantee QoS. Differential services are applied in internet protocol version 4, but different applications such as audio video and streaming would better performed under the integrated services model because it has relatively low cost and bandwidth.

The RSVP can provide end-to-end resource reservation as a control protocol, which is used for the integrated services model. That is RSVP allows Internet real-time applications to reserve resources before they start transmitting data.

However, RSVP is designed for fixed network and it cannot provide guaranteed resource reservation in wireless mobile environment as UMTS. In this article, we provide extensions in RSVP to make QoS guarantee for mobile nodes without affecting the capacity of the network.

II. REVIEW OF THE STATE OF ART

Recently there have been several studies in addressing the problem of supporting mobility on RSVP [1]. Talukdar proposed Mobile RSVP (MRSVP), as an extension to conventional RSVP in which he introduced home agent, remote agent as well as several control messages [2]. Chen proposed a new signaling protocol based on IP multicast for mobile hosts to reserve resources in integrated services internet [3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

From our perspective, there still exist some problems in these solutions in providing real-time services in wireless mobile networks and we would focus on MRSVP. MRSVP use Mobility Specification (MSPEC) to identify the neighbor cells which need to establish advance reservation. Two reservation styles are defined in MRSVP: Active Reservation and Passive Reservation. When a Mobile Station (MS) switches to a new cell from native cell, remote agent in this cell would notify all mobile agents in its neighborhood to establish advance reservation and meanwhile change its reservation style from Passive to Active. However, MRSVP should establish advance reservations in all possible neighbor cells, which waste too much resource. So, how providing QoS guarantees using less resource in the network makes a new challenge to us.

IV. PROBLEM SOLUTION

In order to address the end-to-end resource reservation through cost-effective access, our main extensions in RSVP introduce proxy agent and global position system (GPS). Proxy agent is a default gateway in a cell and it will predict the next cell, which MS will move to based on information of MS’s position and speed measured by GPS. Similar to MRSVP, we also introduce Normal Reservation and Advanced Reservation. However, sender sends RSVP messages and application data packets to MS using Multicast [4].

When MS moves to new cell, the proxy agent in the old cell should release the resource reservation (RS) while the proxy...
FlowSwitch: Proxy agent uses the message to update the RS state and data stream route switch.

ResvTear: Proxy agent sends the message to another Proxy agent in neighbor cell notifying it to release Advance RS for a MS.

Now MS is in cell B, using GPS Proxy agent in cell B can justify that MS will move from cell B to cell C. So Proxy agent in cell B sends a MultiMes message to Proxy agent in cell C, after receiving the message, Proxy agent in cell C adds to the Multicast Group, when PC (sender) sends PATH message next time (because PC sends PATH message periodically), these two Proxy agents can receive this message, so Advance Reservation can be established. At this time, if MS’s moving orientation has changed, Proxy agent in the home cell will notify (send ResvTear message) the Proxy agent in the next cell to release RS.

After process mentioned above, when MS comes to the margin of the cell C then cell switch will happen soon. Proxy agent in cell B sends SwitchNotify message to Proxy agent in cell C, then Proxy agent in cell C will send FlowSwitch message to Router R3, when R3 receives this message, it will change Advance Reservation state to Normal Reservation state, after this R3 will send FlowSwitch to R1, when R1 receives this message, it will change Advance Reservation state to Normal Reservation. At the same time, this process is also happening in cell B—R2—R3, because R1 is the merge point so data packet switch will end up in the point.

In addition to the messages present in RSVP, some additional messages are required in extensions as following:
- MultiMes: MS sends this message to Proxy agent notifying it to add to Multicast Group, the message contains related Multicast address.
- AdRESV: The message is used to distinguish from RESV and to establish Advance Reservation.
- SwitchNotify: The message is used to notify MS’s position switch, it’s transferred between different Proxy agents.

V. CONCLUSION

This paper considered the problem of existing RSVP in providing real time services in wireless mobile networks, while reducing excessive RSVP reservation. We also give description, how integrated services interlinked into a wireless networks. First we introduced two styles of reservations of RSVP. Second, we also introduced proxy agent, GPS as well as some extensions in RSVP messages and we proved it logically worked by showing the flow diagram. However, QoS will fail due to lack of resource to establish advance reservation, this can be left as future research work.

ACKNOWLEDGMENT

We are very thankful to our respectable Professor Wlodek for this kind of help and guideline in completing this paper.

REFERENCES

Abstract—Many professions these days require the use of protective mask in order to prevent all sorts of hazards. But use of protective masks creates disturbances between co-workers when communicating with each other using electronic communication devices e.g. microphones placed inside the masks. The reason is that the speech signal from the mouth is distorted by the environment of the mask. This effect can be reduced by filtering the microphone output with a filter that is the inverse of the mask channel. The main aim of this paper is to find the inverse filter of the mask channel and its characteristics by using Least Mean Square (LMS) algorithm.

Index Terms—LMS Algorithm, Adaptive equalization algorithms, echo, inverse filter.

I. INTRODUCTION

In this paper the aim is to solve the problem faced by many workers while working in hazardous conditions like in a nuclear reactor etc. Any person working in such an environment must protect himself by wearing some protective masks, due to this protective mask channel, communication among the employees becomes difficult.

We employ the adaptive channel equalization using LMS algorithm in addressing this problem. A speech signal from speaker is applied to the microphone and the output is provided for analysis. The applied sampling frequency was chosen 8 KHz to meet the data and bandwidth requirement. For this paper the work on the actual equipment is not done, provided signals are the recordings of dummy equipment. The simulation of the algorithm is carried out in Matlab.

II. REVIEW OF THE STATE OF ART

The selection of adaptive filter structure and algorithms for adaptation are necessary while improving speech quality by echo cancellation [1]. There are various adaptive algorithms like LMS, Normalized LMS (NLMS), Leaky LMS (LLMS) and Recursive Least Square (RLS). The RLS gives the best attenuation and converges much faster than other algorithms but has high computational complexity [2].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

In this paper we describe how to improve speech quality in a gas mask using LMS algorithm. We consider the communication discomfort the workers experience when wearing a protective gas mask against hazardous conditions like working in a nuclear reactor, due to the protective mask channel, which causes distortion of the speech signal from the mouth. The main contributions of this paper are:

- Using LMS algorithm.
- Adaptive Channel Equalization.
- Inverse Filter.

IV. PROBLEM SOLUTION

A. Adaptive Channel Equalization:

Fig. 1. Schematic block diagram of the problem statement

A speech signal \( s(n) \) from the mouth speaker is applied to microphone via mask channel. Because of the mask channel we get the distorted signal \( x(n) \), here our aim is to reduce disturbances produced by the mask channel. This can be done by the inverse filter, placing the inverse filter between the channel and output from the speaker. In order to estimate inverse filter, adaptive channel equalization method was employed.

The role of the adaptive algorithm is to reduce the mean square error (MSE) [3] represented by the following equation

\[
\xi(n) = E\{|e(n)|^2\} \quad (1)
\]

Where the error \( e(n) \) can be evaluated from model as

\[
e(n) = d(n) - y(n) \quad (2)
\]

Where \( d(n) \) is the desired signal. The output from the inverse filter \( y(n) \) can be mathematically expressed as

\[
y(n) = w(n)^T x(n) \quad (3)
\]
where \(\mathbf{w}(n)\) and \(\mathbf{x}(n)\) are the vectors containing inverse filter coefficients and distorted signal samples respectively. Inverse filtering can be solved analytically with the help of Wiener Hopf equation given by

\[
R_{xx}\mathbf{w} = r_{dx}
\]  

(4)

Where \(R_{xx}\) is the autocorrelation matrix of the input signal and \(r_{dx}\) is the cross-correlation vector between the input signal and desired signal.

B. Least Mean Square Algorithm:

The update equation for the Least Mean Square (LMS) algorithm [3] is given by

\[
\mathbf{w}(n+1) = \mathbf{w}(n) + \mu \mathbf{e}(n)\mathbf{x}(n)
\]  

(5)

where \(\mathbf{e}(n)\mathbf{x}(n)\) is the estimate of the gradient of \(\mathbf{x}(n)\) and \(\mathbf{\mu}\) is known as the step size and it is computed as

\[
0 < \mu < \frac{2}{(p+1)E[|x(n)|^2]}
\]  

(6)

Where \(p\) is the length of the filter and \(E[|x(n)|^2]\) is power in the signal \(x(n)\). Choosing appropriate length of the filter is very important in designing of inverse filter. As a rule of thumb a designer has to choose filter length in such way that the MSE should be minimized.

C. Simulation Result:

The delay illustrated in Fig.1 needs to be chosen correctly to minimize the mean square error in this application. Throughout the implementation of algorithm in this paper the delay was considered as half of the filter length.

The figure below shows the plot between the mean square errors versus delay for LMS algorithm, where it can be seen that the MSE is minimized at half of the filter length.

![Fig. 2. MSE Vs delay for LMS is 28 with filter length 56](image)

The plot for Power Spectral Densities (PSD) is shown in the Fig.3. PSD is very important tool to identify oscillatory signals in time series data as it shows the energy variations as a function [4] of frequency that is it shows at which frequencies variations are strong and at which frequencies variations are weak. As the input to the mask channel is white noise, therefore the PSD of the signal \(x(n)\) should mimic the frequency response of the mask, from the plots we can observe that in the case of LMS, ‘y’ is equal to ‘s’ over the frequency range of 0-1500Hz. Therefore we can conclude that the designed inverse filter is the inverse of the mask.

![Fig. 3. Power Spectral Density of Signal with LMS algorithm.](image)

V. CONCLUSION

In this paper it can be seen that the errors produced due to the mask were minimized and the implementation of LMS based algorithm gave a reasonable result with less complexity. This less complexity was achieved at the expense of the slow convergence of the algorithm. We therefore, proposed that for further work on the project an algorithm with less complexity and fast convergence could be implemented.

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We would like to express our sincere thanks and gratitude to Prof. Wlodek Kulesza for being our mentor on this journey.

REFERENCES

Voice Activity Detection Using Feature Vectors

Jagadeesh Thati, Naga Praveen Parchuru, and Yasir Masood Malik

Abstract—Effective speech communication can be achieved by taking the speech signal when microphone is active and suppressing the noise when it is passive. The model we proposed in this paper is to take Feature vectors of pre-defined speech and noise signal’s, which already are stored for processing. Then centroids of the Feature Vectors were calculated using k-means algorithm. The Minimum distance between framed feature vectors of input signal and centroids of pre-defined signals are estimated using Euclidian distance. The ratio obtained between noise and speech minimum distance vectors will represent the voice activity at the microphone. The evaluation of the ratio indicates the significant performance of voice activity detection in noisy environment as well.

Index Terms—Voice activity, Feature vectors, k-means clustering.

I. INTRODUCTION

Effective speech communication is increasingly becoming important in the modern era of telecommunication system. This increase can be largely attributed to the desire to lower the average bit-rate of speech communication systems, whether this is for mobile telephony or VoIP communications [1]. Algorithm for effective speech communication can be developed from various methods specific to different applications. Voice activity detection is one of the methods which can be used to improve the quality of the speech signal, minimizing the channel capacity and suppressing the noise when there is no speech signal.

II. REVIEW OF THE STATE OF ART

Voice activity detection techniques have been formalized for speech enhancement, low bandwidth and minimum error rate while transmission of speech signals. Bayesian adaptive algorithm is used for the lost detection rate and the error rate of transmitted packets [2]. Some other methods were also proposed for the Voice activity detection using wavelet transform [3], in which energy of the sub bands and feature extraction is done. In this process of estimation, the main concentration was only on the parameters, feature extraction and then getting output after fixed threshold.

To enhance and suppress the noise signal in the microphone is the area of research, which has gained much importance in the field of speech processing’s techniques. Some of them are included as the, Standard Voice activity detection algorithms, Hybrid algorithm and PHAT methods.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

The way of extracting the feature vectors of pre-defined signals and input signals, how to determine the voice activity at microphone using these feature vectors. This could be done by extracting feature vectors using Levinson-Durbin algorithm, and then these feature vectors of pre-defined speech and noise signals will be compared to the input signal to detect the speech activity at microphone.

IV. PROBLEM SOLUTION

Robust Voice Activity Detection is a field that is receiving considerable attention because of its relationship, for example, speech recognition [4].

To detect the voice activity at microphone, firstly we need to extract the feature vectors of pre-defined speech and noise signals. In our case we took 4 signals each for speech and noise.

In real time processing it is very hard to take the whole signal to estimate the voice activity at microphone, for this we have to process short frames of the signal to get the quick response. After framing the signals we get 160 samples per frame. Then extracting 10 feature vectors ($I_n$) from each frame of pre-defined signals using Levinson-Durbin algorithm [5]:

$$I_i = [r(i) - \sum_{j=1}^{i-1} \alpha_j^i (i-j)]/E_i$$

Where $\alpha_i$ is:

$$\alpha_i^i = I_i$$

$$\alpha_j^i = \alpha_{j-1}^i - k_i, \alpha_{j-1}^j$$

$$e_i = (1 - r^2)\epsilon_{i-1}$$

$I_i$ is the feature vectors, $\alpha_i$ is the prediction filter and $e_i$ is prediction error, $r(\tau)$ is the autocorrelation of the signal. Length of the feature vectors is dependent upon the order of $r(\tau)$. As 10 feature vectors will be calculated for each frame, so the order of $r(\tau)$ should be 10. $e_i$ is the prediction error which will be minimized during the training of feature vectors.

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These feature vectors now can be used to determine whether the signal coming from a microphone is a speech signal or a noise. For this we have to frame each incoming signal and extracting its feature vector and finding its minimum distance with the centroids of pre-defined speech and noise feature vectors. That minimum distance and centroids of pre-defined feature vectors can be determined using Euclidean distance and K-Means Algorithm [6], respectively.

\[ E = \text{argmin} \sum_{i=1}^{n} \sum_{j=1}^{m} ||x_{ij} - z_i||^2 \]  

(5)

\( E \) is the minimum distance between the input signal’s feature vector and centroids of pre-defined feature vector. \( x_{ij} \) is the feature vectors of input signal and \( z_i \) is the centroid of the pre-defined feature vectors. We will calculate \( k=10 \) centroids each for both pre-defined speech and noise feature vectors. Minimum distance from the input signal’s feature vector with centroids of pre-defined speech is \( d_s \) and minimum distance from the input signal’s feature vector with centroids of pre-defined noise signals will be \( d_n \).

Ratio calculated from \( d_n \) and \( d_s \) is:

\[ r = \frac{d_n}{d_s} \]  

(6)

This shows the peaks when there is a speech signal and valleys when there is noise. Fig 2 shows the same ratio values for pure speech signal as well as the distorted speech signal. This shows that the algorithm is robust in noisy environment. So we can use this ratio to attenuate the noise in the incoming signal from the microphone.

V. CONCLUSION

This paper presented the voice activity detection using feature vectors. In the proposed model, we first framed the signal into blocks, and then extracted the feature vectors of pre-defined speech and noise signals, which were already stored for processing. We used k-means algorithm to calculate the centroids of the pre-defined framed feature vectors. Levinson-Durbin algorithm is used to extract the feature vectors. The minimum distance between framed feature vectors of input signal and centroids of pre-defined signals are estimated using Euclidean distance. The ratio obtained from the Euclidean distances of noise and speech, represents the voice activity at the microphone. The evaluation of the ratio indicates the significance performance of voice activity detection in ideal and noisy environment.

This project paves the way to advancement in the field of Acoustic localization and Voice activity detection techniques, such as effective channel utilization. Moreover this technique has the capacity of working on the bit rate, which can be increased, thus making the channel more efficient.

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REFERENCES

An Improved Version of Filtered XLMS Algorithm in Feed Forward Method
(Approach of Normalizing Step and Leaky Factor)

M.Suryanarayana Murthy, Tulasi Pandra, and Bo Jiang

Abstract—Many Authors have modified the parameters for the (Filtered-X-least mean square) FXLMS algorithm to improve the performance of the Feed Forward method. That is because Feed Forward Method is the most common method in the Active Noise Control. Here in this paper we will improve the performance by normalizing the step size and adding leaky coefficient to overcome the instability requirements.

Index Terms—Active Noise Control, Filtered-X least-mean-square (FXLMS) algorithm, Adaptive Signal Processing.

I. INTRODUCTION

ACTIVE Noise and Vibration Cancelation creates a great intention towards reducing acoustic noise and vibration at low frequencies and it’s more efficient than passive methods i.e., silencers. Adaptive Feed Forward systems are self tuning because they can tune to small changes in the system being controlled. Feedback controllers are non-adaptive. In these methods, estimating the noise signal from the source and generating the anti-noise signal is the principle. Feed Forward method is efficient in controlling noise and vibrations shown in the Fig.1. On the basis of adaptive algorithm, noise cancelation transfer function and the error sensor signal to generate the anti-noise [1]. The performance optimization of the adaptive algorithm is done by considering the convergence coefficient ($\mu$), so called step size and the leaky factor ($\alpha$).

II. REVIEW OF THE STATE OF ART

Several papers have been published based on improving the system performance by modifying the adaptive algorithm in Feed Forward method with variable step size for estimating secondary propagation path [2]. Reduce the effect of measurement noise corrupting both reference and error signal by averaging the weights [3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Filtered XLMS is the extension of the least mean square (LMS) algorithm. In LMS, the reference signal is directly feed to the adaptive algorithm but in Filtered XLMS the reference signal is filtered. There are three ways to improve the LMS algorithm, Long term stabilization, Minimize the residual error and optimizing the tracking speed. It is very hard to improve the algorithm considering with all the three ways together. Our contribution in this paper is to implement the performance of the algorithm using normalizing convergence coefficient and leaky factor at the same time.

IV. PROBLEM SOLUTION

If the input process of the autocorrelation matrix contains zero Eigen values then the adaptive filter modes are unstable. To get long term stability, we introduce the tap leakage i.e. leaky factor to avoid overflow of weight coefficients $w(k)$. However, it increases the mean square of the residual error so that we have to select the optimum value of the leak coefficient, and the weight equation becomes (1).

$w(k + 1) = w(k)(1 - \alpha \mu) + 2\mu e(k)x(k)$

$R' = R + \alpha I$

Where:

$R$ is Autocorrelation of noise signal i.e. $x(k)$
$e(k)$ is error signal

The improvement of the speed of convergence depends upon the convergence coefficient. We select the convergence coefficient by normalizing the original step coefficient by the power of the original signal. Combined with normalizing step and leaky factor the final weight equation (2).

$w(k + 1) = w(k)(1 - \alpha \mu) + \frac{2e(k)x(k)}{\bar{x}(k)\bar{x}(k)}$

By modifying the convergence coefficient and leaky factor, the algorithm is shown in Fig. 2.

Applying the noise signal to the system described in Fig. 1. Then the simulation results from the Matlab are the error signal performance is shown in Fig. 3. The noise and the improved attenuated noise power spectral densities are shown in Fig. 4.
V. CONCLUSION

In this paper, the Improved Filtered XLMS algorithm enhanced the speed of convergence and stability of the system, which filters the reference signal by the estimated Forward Path. The Simulation Results show the improved performance compared to the normal algorithm of the active noise control system.

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REFERENCES

Matlab Toolbox for Link Budget Analysis in Communication Networks

Christopher Ojaide, Hasnain Kashif, and Usman Shafique

Abstract—Link budget calculations are essential in the planning of communication networks. When a signal is transmitted from a transmitter to a receiver, the received signal power varies randomly. The transmitted power is degraded gradually due to losses such as path, rain, and cable losses as well as Additive White Gaussian Noise (AWGN) and Rayleigh fading. In this work, we built a Matlab toolbox that helps to examine the effect of link budget loss factors. The toolbox is intended to be used by students and researchers in the field of communication networks. The toolbox shows the output of the signal that is being analyzed after assigning chosen values to the loss factors.

Index Terms—AWGN, link budget, path loss, rain loss, Rayleigh fading.

I. INTRODUCTION

COMMUNICATION networks such as mobile telephony, satellite communications and Bluetooth connectivity is very important in our daily life. Therefore, the estimation of the transmitted signal power in the radio channel, and the achievement of appropriate signal coverage area with a satisfied quality of signal strength has become important in communication systems for private and commercial purposes. A link budget is a determinant of the strength and quality of signal that is received at the receiving end of the communication system. Therefore a researcher, student or design engineer working in the field of communication engineering needs a proper analysis of the link budget in order to make proper decisions in their work.

II. REVIEW OF THE STATE OF ART

Link budget analyzes signal loss factors during signal propagation and estimates the required power from the transmitter (Tx). It is the calculation of all the gains and losses that occurs during signal transmission through mediums such as free space, fiber, cable etc. to the receiver (Rx) in the communication system. It calculates the attenuation of the transmitted signal due to propagation losses and antenna gains.

A simple link budget equation is:

\[ P_{RX} [\text{dBm}] = P_{TX} + \text{Gains} - \text{Losses} \]  

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Two important questions are how to estimate the received signal losses caused by the propagation loss factors and how these losses affect the received signal.

IV. PROBLEM SOLUTION

By using link budget model to estimate the receive signal level, it becomes possible to predict the SNR for the communication system [2]. In radio channel, received power can be obtained by adding gains and subtracting losses.

\[ P_{RX} = P_{TX} + G_{TX} - L_{TX} - L_{FS} - L_{M} + G_{RX} - L_{RX} \]  

Where:

- \( P_{RX} \) received power [dBm]
- \( P_{TX} \) transmitter output power [dBm]
- \( G_{TX} \) transmitter antenna gain [dBi]
- \( L_{TX} \) transmitter losses (coax, connectors etc.) [dBm]
- \( L_{FS} \) free space loss or path loss [dBm]
- \( L_{M} \) miscellaneous losses (fading margin, body losses, polarization etc.) [dBm]
- \( G_{RX} \) receiver antenna gain [dBi]
- \( L_{RX} \) receiver losses (coax, connectors etc.) [dBm]

We developed a Matlab toolbox based on (2) that can be used to predict and analyze the received signal as shown in the process flow chart of toolbox in Fig.1.

![Fig. 1. Toolbox process flow chart.](image-url)
The toolbox user can introduce different loss factors on the signal to be transmitted and examine the received signal by clicking the Apply button. The toolbox has different parameters and their values can be changed.

**Signal module**: The frequency [MHz] and amplitude [volts] of the signal can be set to any value.

**Antenna module**: It can be set to dipole, biquad, helix and parabolic with gains at 2, 8, 15 and 24 respectively.

**AWGN module**: It adds white Gaussian noise to signal. SNR is the signal to noise ratio per sample in [dB].

**Miscellaneous module**: This is made up of Tx/Rx, connector and combiner losses. It ranges from 2 to 10 [dB].

**Rayleigh fading module**: It is a frequency-flat Rayleigh channel loss that results from the Doppler frequency [Hz] of pedestrian, bicycle, medium speed car and fast moving car with a velocity of 4, 15, 80 and 120 [km/hr] respectively.

**Rain loss module**: It is 3.3 [dB/km]. It can be set to any chosen kilometer.

**Path loss module**: This loss depends on distance and path loss exponent and can be set to free space, urban area cellular, shadowed urban, in building line of sight, obstructed in building and obstructed in factories.

Fig. 2 shows the main interface while Fig.3 and Fig.4 shows the signal output after applying losses.

V. **CONCLUSION**

A Matlab toolbox that is meant for the analysis of link budget for communication networks was designed. The toolbox is very easy and convenient to use. This study presents both theoretical and practical significance. Future research will be based on link budget for Worldwide Interoperability for Microwave Access (WiMAX) or Long Term Evolution (LTE) with respect to radio resource management.

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**REFERENCES**


Dynamic Channel Allocation Strategy using Reserve Channel respect to Handoff Failure Improvement

Zeeshan Feroze, Roland A.Ogbokor and Naresh R. Banda

Abstract—In Telecommunication sector, we are been faced with many constraints, of which handoff failure is one of them. Handoff failure has become very disturbing because most users experienced forced termination of their calls while the call is ongoing. As population grows up, the number of users also increases tremendously. Cells size has also been reduced to meet this demand. We proposed in our paper that more concentration should be given to handover calls rather than new incoming calls. This is a way of creating better Quality of Service (QoS) for users. We proposed here Reserved Channel Scheme (RCS) to handle this problem. We also used MATLAB as a tool for simulation.

Index Terms—Handoff, Reserved Channel Scheme (RCS)

I. INTRODUCTION

In this present era, wireless communication has increased rapidly and in the same way, cellular users are more. More people find it easier access of communication. The capacity of the system is increased by reusing the resource more intensively in high traffic demand area. While a call is ongoing, efforts are made to ensure the call continuation as users move from one coverage cell to another. Handoff failure has to be minimized to give users certain measure of confidence while the call is ongoing. A reserved channel scheme is implemented, which shows a reduction in the rate of handoff failure.

II. REVIEW OF THE STATE OF ART

In minimizing handoff failure, a modified version of dynamic channel allocation called compact pattern with maximized channel borrowing (CPMCB) was implemented. This outperform borrowing with channel ordering (BCO), borrowing with directional channel locking (BDCL) in a fixed channel allocation (FCH) [1]. It was also achieved by applying measured-based priority scheme (MBPS) [2]. Network monitors the power levels of a queued handoff call dynamically via the portable which monitors the signal strength of the radio link and the poorest in the queued handoff is given highest priority [2]. Handoff failure is also reduced by increasing the number of reserved channels.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

In this experiment our intention was to minimize the probability of handoff failures in personal communication network (PCN) by introducing reserved channel scheme. Our main approach to the problem is via allocating 25% of the total channels to be the reserved channels. Allocation of less reserve channels to low density areas when compared to high density areas with respect to users. Our main statement was proved by MATLAB simulation and can be seen in the figures.

IV. PROBLEM SOLUTION

This is the algorithm used for the reserved channel scheme. This is a case of uniform traffic. $S$ is the total number of channels in the cell. $C_h$ is the reserved channels allocated only for handoff calls ($S-n$). While $n$ are the channels allocated for both new incoming calls and handoff calls. Rho ($\rho$) is the offered traffic load in Erlang /cell ($\rho=\lambda_{in}/\mu$). $\mu$ is the rate calls are leaving the cell and $\eta$ is the rate handoff calls are leaving the cell.

![Fig.1. Algorithm of the Reserved Channel Scheme](image-url)

It is assumed that, new call attempts in a cell follow a Poisson process. $\lambda_{in}$ Represents the new call arrival rate and $\lambda_{out}$ represents the handoff call arrival rate to a given cell. $P_j$ Be the statistical-equilibrium probability of $j$ busy servers.

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In Fig. 2 the flow chart is of reserved channel scheme. Any incoming handoff call has to be allocated a free channel to continue ongoing call. If not, the call will be forced to terminate.

**V. CONCLUSION**

Simulation was done using MATLAB to study the probability of handoff failure in a reserved channel scheme. We observed that, when 25% of the total channels were given as reserve channels for different cells capacity, the cell that has the largest reserve channels proved to have the minimum value of handoff failure probability.

Future research work can be achieved in the reserve channel scheme, where handoff calls can be queued in a manner that some should be given more priority than others while trying to gain access to a free channel during handoff.

**REFERENCES**


Using MUSIC Algorithm for Extraction of Fetal Electrocardiogram

Azhar Ali Mian, Seyedeh Somayeh Hosseini, and Sardar Usman Khalid

Abstract—The paper presents the use of Multiple Signal Classification (MUSIC) Algorithm to obtain Fetus Electrocardiogram (FECG) from corrupted ECG record on mother’s stomach. The ECG spectrum contains three frequency contents mother’s heartbeat, fetus’s heartbeat and noise from power supply, by using MUSIC algorithm’s functionality of spectrum estimation, it is tried to separate those frequency contents and consequently extract fetus’s heartbeat. The idea is validated by the help of a real life example.

Index Terms—Multiple Signal Classification MUSIC, Spectrum Estimation ES, Electrocardiogram ECG.

I. INTRODUCTION

MUSIC algorithm is a promising tool to detect Direction of Arrival (DOA) in multiple transmitter single receiver scenarios, another effective use is estimation of spectrum. It finds the spectra by using an eigenspace method. Considering its popularity and effectiveness in DOA detection an effort has been made to test its usefulness in separation of signals containing low frequencies. This paper is optimized on the facts that what are the consequences and results, while applying MUSIC algorithm for the separation of signals?

There have been many efforts made prior to this, to separate the signals by using different methods. This paper is an effort to illuminate the merits and drawbacks while applying MUSIC algorithm over a scenario which is different from this algorithm’s definition.

II. REVIEW OF THE STATE OF ART

FECG extraction from the corrupted ECG on mother’s stomach is basically an Adaptive Signal Processing problem. Two different methods are currently being used, Blind Source Separation (BSS) [3] using Independent Component Analysis (ICA) and Adaptive Filters [2].

ICA [3] is a way to find the independent components of a multivariate random variable. These components are directions in which elements after random variable have no dependency.

Assume the following basic linear statistical model:

\[ Y = MX + N \]

in which \( Y \) is referred to as the observation vector, \( X \) is called the source vector and \( N \) represents additive noise. \( M \) is the mixing matrix.

The goal of ICA now consists of the estimation of the transfer matrix \( M \) and/or the corresponding realizations of the source vector \( X \); given only realizations of the output vector \( Y \):

- The columns of \( M \) are linearly independent;
- The components of \( X \) are mutually statistically independent, as well as statistically independent from the noise components.

On the other hand by using adaptive filters [2] the unwanted contents are filtered out and desired ECG is extracted. In this case Averaging method and Multiple Reference method are two approaches. The adaptive algorithm estimates mother ECG and subtract it from the input to extract fetus ECG as output. LMS Least Mean Squares, NLMS Normalized Least Mean Squares, LLMS Leaky Least Mean Squares and RLS Recursive Least Squares [1] are the algorithms that are used to find the optimal coefficient of the filter based on least mean square of the error signal (difference between desired and the actual signal).

III. RESEARCH QUESTION AND PROBLEM STATEMENT

MUSIC estimates the spectrum of a signal using eigenspace method and DOA. It separates noise subspace from signal subspace assuming that noise is uncorrelated with signals.

Due to the fact that MUSIC can just separate uncorrelated contents, it is needed to find a way to improve this algorithm so that it will be able to separate correlated contents as well.

How to apply MUSIC to extract FECG?
Which technique should be used with MUSIC to separate the signals?
Contributions:
- To help improve biomedical devices
- To find out the best suitable technique to be used with MUSIC

IV. PROBLEM SOLUTION

A. Method

MUSIC estimates the frequency content of a signal-column vector or autocorrelation matrix using an eigenspace method. It assumes that a signal, \( x(n) \), consists of \( p \) complex exponentials in the presence of Gaussian white noise. Given an \( M \times M \) autocorrelation matrix, \( R \), if the Eigen values are
sorted in decreasing order, the eigenvectors corresponding to the p largest Eigen values spanning the signal subspace. The general idea is to use averaging to improve the performance of the Pisarenko’s estimator [2].

The frequency estimation function for MUSIC is

$$\hat{P}_{MU}(e^{j\omega}) = \frac{1}{\sum_{i=p+1}^{M} |e^{j\omega}v_i|^2}$$

(1)

Where V_i are the noise eigenvectors and

$$e = [1 \ e^{jw} \ e^{j2w} \ ... \ e^{j(M-1)w}]^T.$$  

(2)

B. Validation

The input is a matrix containing eight vectors. The first five vectors are different realizations of mother ECG, fetal ECG and 50 Hz of noise and last three vectors are realizations of mother ECG.

pmusic() is a MATLAB functions which applies MUSIC algorithm to estimate pseudo spectrum of the given signal, this pseudo spectrum is calculated using estimates of Eigen vectors of the correlation matrix associated with the input.

The output graphs derived after using pmusic() are illustrated below in Fig. 1 and Fig. 2

![Fig. 1 Spectrum Estimation of Mother and Fetus ECG](image1)

![Fig. 2 Spectrum Estimation of Mother and Fetus ECG](image2)

The heart beat frequency of mother and baby is in the range from 1Hz to 2Hz.

Maximum frequency for all humans is 2Hz, 50Hz noise is included in the input signal, so the maximum frequency content F_{max} is 50Hz, Nyquist rate F_S = 2 F_{max} means sampling frequency must be 2*50 = 100Hz.

In this case the F_s is 500Hz, therefore the F_s fulfills the requirement.

There is a separate heart beat record of mother, so it is already known which peaks correspond to mother heart beat, then other peaks correspond to baby heart beat, using a suitable filter FECG can be extracted.

V. Conclusion

In this paper while applying MUSIC algorithm on the ECG obtained from pregnant woman’s stomach it is observed that, MUSIC algorithm is unable to fulfill the requirement to separate FECG from MECG and other frequency contents.

However, it helps in eliminating the noise subspace from the input signal and returns spectrum estimation for MECG and mother plus fetal ECG.

Hence, MUSIC algorithm independently can’t be used to separate given input, however using frequency estimation from application of MUSIC algorithm a new filter can be designed to separate these signals.

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REFERENCES


Implementation of DSR, AODV and OLSR in Wireless Mesh Networks: A comparative study

Jason W. Mwanza, Member, IEEE, Kashif A. Muhammad, and Imtiaz A. Yousafzai

Abstract—Wireless Mesh Networks (WMN) are self-organizing, however, sudden changes in the network topology due to the movements of mobile nodes result in increased latency. A good choice of a routing protocol in the implementation of WMN significantly reduces this latency. This paper discusses the implementation of three different routing protocols: Dynamic Source Routing (DSR), Ad hoc On-Demand Distance Vector (AODV) and Optimized Link State Routing Protocol (OLSR) in Mobile Ad Hoc Networks. OLSR builds consistent paths with lower hop count compared to the reactive protocols such as AODV and DSR. This reduces end-to-end latency and improves overall network performance in WMN.

Index Terms—Mobile Ad Hoc Networks, Ftp, Routing protocols

I. INTRODUCTION

There are a variety of scenarios in which Wireless Mesh Network (WMN) is deployed. In this research, we considered Wireless mesh in Mobile Ad hoc Network (MANET). A mesh network is reliable and offers high redundancy. It is easy to configure a MANET and the construction is inexpensive since the movement of nodes in the network are not restricted and do not require central administration [1].

An investigation of the three main routing protocols implemented in WMNs was performed by comparing latency, network load and throughput offered by these protocols. This investigation was carried out with the help of a simulation tool: Opnet Modeler 11.0. The results in this study clearly spell out the need for a good choice of a routing protocol in the implementation of WMN.

II. REVIEW OF THE STATE OF ART

There are several competing schemes for routing traffic across mesh networks. In this paper the three main schemes are discussed: Dynamic Source Routing (DSR), Ad hoc On-Demand Distance Vector (AODV) and Optimized Link State Routing Protocol (OLSR).

DSR uses source routing and it does not rely on the routing table at each intermediate device. Determining source routes requires accumulating the address of each device between the source and destination during route discovery. The source routing nature of DSR increases the overhead when the network is increased in size [2].

AODV Routing protocol uses an on-demand approach for finding routes. A route is established only when it is required by a source node for transmitting data packets. It employs destination sequence numbers to identify the most recent path. ‘AODV uses a table-driven routing framework and destination sequence numbers’ [3].

OLSR protocol uses Hello messages at each node to discover 2-hop neighbor information and performs a distributed election of a set of multipoint distribution relays (MPRs). Nodes select MPRs such that there exists a path to each of its 2-hop neighbors via a node selected as an MPR. The paths that OLSR builds are consistent with lower hop count compared to the reactive protocols such as AODV and DSR. This reduces end-to-end latency and improves overall network performance [4].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

With more than 70 competing routing protocols, the choice of the most efficient one has become an important factor to consider when deploying a WMN. The main question in this research is which routing protocol, among DSR, AODV and OLSR, best enhances network performance in WMNs.

Our hypothetical assertion is that OLSR offers high network performance in terms of end-to-end delay, network load and throughput in WMNs compared to DSR and AODV. This study was performed to ascertain the assumption made in our hypothesis and answer the questions that surround the choice of a routing protocol.

IV. PROBLEM SOLUTION

In this thesis we used File transfer protocol (Ftp) traffic to analyze how implementing a specific routing protocol in WMN affects network performance. Three main WMN routing protocols: DSR, AODV and OLSR were modelled.

A. Simulation Environment

We set up a mobile Ad Hoc network simulation environment as shown in Fig. 1 below. 25 Mobile nodes randomly discovered paths to the server. Total link bandwidth was set to 1Mb/s.
Simulation environment
Mobile node: 25
Service: FTP
Routing protocol: DSR, AODV and OLSR

We injected ftp traffic in the network for all the three scenarios at an average rate of 0.116296 Packets/s. We measured the performance when mobile Ad Hoc network provides FTP Server. The mobile nodes reorganized themselves based on the routing information and set different paths to the server during the simulation run as shown (Fig. 2) below.

B. Analysis

As shown in Fig. 3 and TABLE I, there was delay of 0.0198s, 0.0142s and 0.0018s in DSR, AODV and OLSR respectively. Despite injecting same amounts of traffic in the three scenarios, it showed that the load was more on the network where OLSR was implemented as compared to the networks with DSR and AODV. In addition, OLSR network presented high throughput followed by the network with AODV then the one with DSR. All the readings were taken with high precision at 99% Confidence Interval (CI).

V. CONCLUSION

This thesis implemented DSR, AODV and OLSR in MANET in three different scenarios. In all the three scenarios we injected equal amounts of Ftp traffic in the network. The results in this study confirm that OLSR offers high network performance in WMNs, compared to DSR and AODV, in terms of end-to-end delay, network load and throughput. The behavior of Transmission Control Protocol (TCP) and User Datagram Protocol (UDP) over MANET can be evaluated in further research to confirm the superiority of OLSR over DSR and AODV.

REFERENCES

The Improved Initialization Technique for Vector Quantizer

Muhammad Ajmal, Sandya Rondla, and Tassadaq Hussain

Abstract—Generalized Lloyd Algorithm (GLA) is a simple and prevalent algorithm for designing a Vector Quantizer (VQ). To solve the problem of empty cell and choice of initial codebook that lead to minimize the average distortion in GLA, we propose an improved initialization technique where average distortion can be dramatically decrease with decreasing clusters size.

Index Terms—Vector quantization

I. INTRODUCTION

The growing amount of information that users wish to communicate or store requires some form of compression for efficient, secure, and reliable use of the compression or storage medium. For example, digitized speech or image generates bit rates too large for many communication links or storage devices. Therefore data compression will be required, which entails a loss of fidelity or an increase in distortion. Hence a fundamental goal of data compression is to obtain the best possible fidelity for a given rate. According to Shannon rate distortion theory results, a better performance can always be achieved by coding vectors instead of scalars, even if the data source is memory less. Based on Shannon results, several design techniques for vector quantizer (VQ) were developed. Among these techniques, Generalized Lloyd Algorithm (GLA), which is based on clustering technique, is one of the most prevalent algorithms. GLA minimizes the average distortion between the training vectors (input data) and assigned code vector by performing several iterations.

The iteration consists of two phases: The codebook assignment phase and the codebook adjustment phase. During the former, each of the training vectors is assigned to the nearest code word in the codebook. While in the later phase, each of the codebook vectors is replaced with the centroid of all the training vectors assigned to that code vector. The process is convergent and minimizes the mean-square error (MSE) of quantizer.

II. REVIEW OF THE STATE OF ART

Several techniques for choosing the initial codebook have been explained in different papers such as splitting technique with preference based on cell population or selection of first M vectors from training set [1], modified K-Mean Algorithm (MKMA) [2], and maximum distance initialization technique [3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

There are two problems in GLA. The first problem is empty cell, which means if any code vector in the codebook gets no training vectors during the assignment phase, then it will not move during the adjustment phase and probably will remain empty during all the iterations that lead to an increase in the average distortion. The second problem is the choice of the initial codebook. As different initial code vectors, in the codebook assignment phase, may result in different fidelity as it will iterate to minimize local distortion [1]. Also, for a desire/given MSE what will be the best block size and codebook size M? Any improved method for initial codebook can also solve the problem of empty cell.

IV. PROBLEM SOLUTION

To solve the problem of optimal initial codebook and best block size for a given MSE value, we proposed a technique based on the innovation of “maximum distance initialization technique” [3]. The procedure can be stated as follows:

1. Calculate the norms of all vectors in the training vectors set.
2. Arrange all the training vectors in ascending order according to their norm values.
3. Form clusters of cells by combining those training vectors whose norm values difference is less than or equal to $d_0$.
4. Compute the centroids of the clusters and set them as code vectors for the VQ.
5. Measure the distortion, by calculating the MSE between the training vectors and Code vectors.

From step 3, we can observe that the size of the codebook may vary with value of $d_0$, which acts as a Euclidean radius. Setting $d_0$ value smaller, we can make MSE smaller i.e. better approximation of the training vectors, but at the cost of increasing the value of M and vice versa. The maximum value of M will be equal to the numbers of training vectors when $d_0$ is equal to zero.

Note that in step 3, when the norms difference between the reference training vector and the comparing training vector exceed the value of $d_0$, then that (comparing) training vector will be set as a new reference vector and this process continuous up to the last training vector so we need only once
the norm comparison of every training vector with the reference training vector. Based on this observation, for given \( N \) numbers of training vectors we will perform \( N \) norm comparison calculations, and as a result this technique has a complexity of \( O(N) \) which is independent of codebook size. While the maximum distance initialization technique has a complexity of \( O(MN) \), for designing of a codebook of size \( M \). Furthermore, our initialization technique has some interesting properties. First the size of the codebook is not fixed and for a desired MSE we can choose an appropriate codebook size. Second, each cell will get some training vectors during the norms comparison step, so it will solve the empty cell problem.

V. SIMULATION

To provide a performance analysis, we took well known images of Lena and baboon from USC database, and applied the modified initialization technique to a set of training vectors, obtained from the pixel intensity coefficients of the images. We took the size of 128 x 128 pixels and 256 grey scales per pixel, examined for different block size of 2 x 2, 4 x 4, and 8 x 8 to find the norms of these training vectors. We set different values of \( d_0 \) and observed that for different values of \( d_0 \) we have different numbers and sizes of clusters. The size of the codebook will be equal to the number of clusters, while the centroid for each cluster will be the code vector for that cluster. To calculate the MSE we used the formula, which is defined as:

\[
MSE = \left( \frac{1}{128} \right)^2 \sum_{i=1}^{128} \sum_{j=1}^{128} (x_{ij} - y_{ij})^2
\]

Where \( x_{ij} \) is the intensity value of the original image at the coordinate \((i,j)\) and \( y_{ij} \) is the intensity value of the quantized image at the coordinate \((i,j)\). Fig. 1 and Table I show the result of simulation. The MSE increases exponentially with \( d_0 \) forms a convex function.

Furthermore, for values of \( d_0 \) below eight, we see that MSE for 8 x 8 size decreases as compared to other block sizes, while for higher value of \( d_0 \) it increases more rapidly. This means that in 8 x 8 size training vectors the adjacent vectors are more correlated as compared to 4 x 4 and 2 x 2 sizes, which decrease as we move further. In Table II, we compared our technique with the maximum distance initialization technique [3] for baboon image and observed that our method provides a better codebook with lower MSE values for all image sizes.

VI. CONCLUSION

An improved initialization technique for the generalized Lloyd algorithm was proposed in this paper which will solve the empty cell problem. Moreover, we developed a relation between \( d_0 \) and MSE for different training vector sizes. This will help us in choosing optimal codebook of size \( M \) for a required MSE value. The proposed technique can also be check for moving images with different resolution and video applications. This technique can give useful results for video applications if the coefficients transform stage does not decorrelate the intensity coefficients values completely e.g. the wavelet transform.

REFERENCES


Fig. 1 Mean Square Error as a function of distance

<table>
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<th>( d_0 )</th>
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<th>( 4\times4 ) block size</th>
<th>( 8\times8 ) block size</th>
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Table I Performance Comparison with Modified Initialization Technique

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Implementation of DCT for Image Compression & Aspect of SNR for different Compression Coefficients

Salma Rayees, Bitra Sridhar and Wang Cang

Abstract—There are number of approaches for compressing image. However, Discrete Cosine Transform (DCT) is often used in image compression. In this paper the DCT-based image compression method is used to achieve lossless compression of two-dimensional images. Here in our research we implement a DCT algorithm for compressing and reconstructing the image at different coefficients, which is being simulated on MATLAB. Distortion is introduced when image is transformed using DCT for which SNR is estimated. So, we calculate the SNR of the image by taking block-by-block of the reconstructed image at different compressed coefficients, which will be useful to determine the acceptability of the compressed image.

Index Terms—Image Compression, DCT, IDCT, SNR

I. INTRODUCTION

COMPRESSION of digital image has been the focus of much recent research it makes possible to use digital transmission and storage by which the raw image data will occupy more memory space, where as compression enables more effective use of transmission and storage resources.

Image compression is the representation of an image in digital form with maintaining an acceptable level of image quality. Many compression techniques have been already proposed in image compression [2]. Here in our research we implement DCT for compression of image and IDCT (Inverse Discrete Cosine Transform) for reconstruction of the image at different compression coefficients using simulation in MATLAB and to know the aspects of SNR for the reconstructed image to its original image.

II. REVIEW OF THE STATE OF ART

The compression of the image by DCT and reconstructed using IDCT at some selected coefficients image is reconstructed [1] is simulated in MATLAB. Quality of the image is observed by SNR which is calculated for reconstructed image [3], comparing to the original image at all coefficients.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

In this paper we have written an algorithm for image compression using DCT and to reconstruct the image using IDCT. This is done using MATLAB simulation with this the quality of the image to its compression coefficients are observed and to know the aspects of SNR for the reconstructed image.

IV. PROBLEM SOLUTION

A. System model

The implementation of the proposed DCT image compression and reconstruction uses DCT transform.

\[ D(i,j) = \frac{1}{\sqrt{2N}} e\left(i\right) e\left(j\right) \sum_{x=0}^{N-1} \sum_{y=0}^{N-1} p(x, y) \cos \left[ \frac{(2x+1)i\pi}{2N} \right] \cos \left[ \frac{(2y+1)j\pi}{2N} \right] \]

(1)

Where \( N \) is the size of the block \( x \) and \( y \) range from 0 to 7, the equation calculates the entry \( (i, j) \) of the transformed image form the pixel value of the original image matrix.

The SNR of the original image with respect to the reconstructed image is calculated by applying the below equation.

\[ SNR = 10\log \left[ \frac{\sum_{x=0}^{N-1} \sum_{y=0}^{N-1} p(x, y)^2}{\sum_{x=0}^{N-1} \sum_{y=0}^{N-1} [p(x, y) - D(x, y)]^2} \right] \]

(2)

Where \( p(x, y) \) is the original image and the \( D(x, y) \) is the reconstructed image.
B. Simulation in MATLAB

The image compression and reconstruction with SNR was implemented in the below algorithm.

*Algorithm:*

a) Convert the image into 8x8 matrix form \( p(x, y) \).

b) Transform the matrix \( p(x, y) \) to its DCT form \( D(i, j) \) applying equation (1).

c) Each block is compressed through quantization.

d) IDCT is applied to the compressed \( D(i, j) \) matrix for all compression coefficients.

e) Calculation of SNR for the reconstructed image using equation (2).

C. Simulation Results

The original image is applied to the simulator gives an output of black and white image with pixels ranging from 0-255 is obtained as the compressed DCT of the original form. The image comprises hundreds or even thousands of 8x8 blocks of pixels during this process.

![Original Image](image1)
![8x8 DCT Compressed Image](image2)

Fig. 2. Original image and compressed DCT image.

The image is reconstructed by using Inverse discrete cosine transform by decoding the bits represent the quantized matrix, each element is multiplied by the original quantization matrix then we get reconstructed image and the image is restored for the coefficients \([1 3 5 10 15 20 30 40]\) by simulating, the result as shown in figure 3, changes can be noticed by the quality of the image varying with coefficients.

![restored image 1 coeffs](image3)
![restored image 3 coeffs](image4)
![restored image 5 coeffs](image5)
![restored image 10 coeffs](image6)
![restored image 15 coeffs](image7)
![restored image 20 coeffs](image8)
![restored image 30 coeffs](image9)
![restored image 40 coeffs](image10)

Fig. 3. Original image and restored image with 1, 3, 5, 10, 15, 20, 30, 40 coefficients.

The SNR is simulated which calculates the ratio between reconstructed image and to its original image. The energy of the image and the energy of the noise is calculated, we get the SNR, the simulation result as shown in graph 1 for the coefficients of the compressed image and to SNR in dB.

![Graph 1. SNR vs. No of Reconstructed Image coefficients.](image11)

V. Conclusion

In this paper, we implemented DCT algorithm for image compression and reconstruction which provides less complexity and small number of computations, which is being simulated on MATLAB for the given image, compressed DCT image is obtained and it is being reconstructed for all compression coefficients by observing this, user can determines the quality of the image and select the coefficients.

The SNR of the image is simulated for the reconstructed image and the original image, by observing the result we can say that the increase in SNR shows that the quality of the reconstructed image is increasing with the amount of compression coefficients.

The future work will be done by comparing all the image compression methods with the same image and to find the best method, which has a high quality image with less compression coefficients.

Acknowledgment

We would like to thank Wlodek J. Kulesza, Professor, Department of Signal Processing, Blekinge Tekniska Högskola, Sweden.

References


Fluid Model Analysis and Application of Wired Personal Area Network
D. Hou, B. Majid, and S. Yousaf

Abstract—The paper investigates well known fluid model which is widely used in communication. By analyzing the relationship between probability of buffer overflow and number of sources/max transmission rate, a simplified formula is derived for fluid model using Least Square Method (LSM). This model is applied to wired backbone Personal Area Network (PAN). Simulation results show that differences are negligible and simplified equation can be accepted.

Index Terms—Personal area network, least square method, fluid model, buffer

I. INTRODUCTION

NETWORK traffic has great importance in today’s demanding and dynamic technology, and network traffic flow has great impact in end to end systems. Analyzing network flow is a key factor towards successful networks design as we are encircled around IP traffic.

Current telecommunication applications have several major dependencies on network traffic as it adversely affects the whole application system services. In order to deal with severe consequences of the traffic flow for deploying and implementing new application services, we need to be accurate in choosing the correct model and statistics for the system. Emerging technologies and applications require higher data rates, guarantee Quality of Services (QoS) and best traffic flow with respect to data handling systems (DHS). We consider the continuous time markov process fluid model for these applications, which derives certain equations to observe network behavior. In [1] and [2], multiple equations provide actual simulated results but have great complexity. Our main work will involve reducing complexity and providing a much simpler equation that shows acceptable results for wired backbone Personal Area Network (PAN). Matlab base simulation results validate our analysis.

II. REVIEW OF THE STATE OF ART

We acknowledge the efforts done in [1] and [2]. It provides detailed observable analysis of the fluid model which is required for the DHS, it further depicts the relationship between buffer overflow probability and buffer size in terms of the number of sources. In [3], the fluid model is applied in home networks that help to analyze existing model to formulate a new model.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

- How to simplify the survival function equation of the fluid model that represents the probability of buffer overflow?
- What is the difference between simulation results of existing probability of buffer overflow equation and new simplified equation?

Hypothesis: Is there a simple equation in the fluid model that represents the probability of buffer overflow.

We have proposed a simple curve fitting formula that has approximately the same results when neglecting minor variations in simulation results.

IV. FLUID MODEL ANALYSIS AND APPLICATION

From [2] we sketch the fluid model depicted in Fig. 1 produced by $N$ sources, transferred to a buffer with a buffer size $x$, and consumed by the receive system continuous in time. Each source is independently exponentially distributed between ‘on’ and ‘off’ states and maximum transmission rate $c$ is defined by $c = C/A$, where $C$ is the channel capacity in buffer and $A$ is a source rate during ‘on’ state. From [1] we obtain final equation for probability of buffer overflow $G(x)$ as

$$G(x) = - \sum_{i=0}^{N-[c|-1]} a_i e^{z_i x} (\tilde{1} \phi_i)$$  (1)

Where $[c]$ denotes the integer part of $c$, $z_i$ is the eigenvalue, $\phi_i$ is corresponding eigenvector, $\tilde{1}$ is unit column vector and $a_i$ is a coefficient obtained from the boundary conditions [1].

Equation (1) provides relationship between buffer overflow probability $G(x)$ and buffer size $x$, but the equation is complex, time consuming and difficult to solve. So it’s necessary to study the relationship and find a simplified formula. However, application of the model is also needed to validate the formula.

A. Fluid Model Analysis

Fig. 2(a) illustrates the buffer overflow probability vs. buffer size. With increase in buffer size, logarithm overflow probability $\log_{10} G(x)$ decreases linearly which approximates to the inverse proportion of buffer size $x$ and is defined as:

![Fig. 1 The structure of fluid model.](image-url)
Eq. (1) and its simulation results are compared with curve fit results in terms of $N$, $c$. Simulated slope results compared with curve fit results in terms of $c$.

$$\log_{10} G(x) = k x$$

(2)

The coefficient $k$ is the slope of the straight line based on parameters $N$ and $c$. We formulate the equation by replacing $k$ with a slope function $f(N, c)$ to get $G(x)$. Equation (2) then can be rewritten as:

$$G(x) = 10^f(N,c)x$$

(3)

Fig. 2(b) illustrates the relationship between slope $f(N,c)$ and the number of sources $N$ when $c$ is fixed. Simulation results show that there exists a ‘cut-point’ for each specific maximum transmission rate $c$. The value of the slope reaches to zero when $N$ exceeds the value of ‘cut-point’, while buffer overflow probability reaches to one. Least Square Method (LSM) is used to fit the simulation data in the non-zero regions using quadratic equation defined as:

$$f(N) = aN^2 + bN + d$$

(4)

The constants $a$, $b$ and $d$ are obtained using LSM.

Fig. 2(c) illustrates the relationship between slope $f(N,c)$ and the maximum transmission rate $c$ when $N$ is fixed. We also obtain ‘cut-point’ for each specific $N$, we fit the simulation data by using linear equation of LSM.

$$f(c) = ec + h$$

(5)

The constants $e$ and $h$ are also obtained using LSM.

The two variables $N$ and $c$ in Equation (4) and (5) have some relationship and can be combined together to form another function:

$$f(N,c) = (e_1c + h_1)N^2 + (e_2c + h_2)N + d$$

(6)

Equation (6) possesses the characteristics of (4) and (5). By fixing $c$, we get quadratic equation (4) in terms of variable $N$, and by fixing $N$, we get linear equation (5) in terms of variable $c$. By Simulating equation (6) we obtain optimal values of the constants, $e_1 = 0.377 \times 10^{-4}$, $h_1 = -9.28 \times 10^{-4}$, $e_2 = -0.003$, $h_2 = 0.098$ and $d = -1.785$. By substituting (6) in (3), we get the simplified function for probability of buffer overflow.

$$G(x) = 10^f(0.377c-9.28) \times 10^{-4}N + (+0.003c+0.098)N-1.785x$$

(7)

B. Fluid Model Application in wired Personal Area Network

To validate simple formula, we apply it in wired backbone PAN. From [3], mean video source rate $A$ is 4.85 Mbps, wired service rate $C$ is 85 Mbps and $N$ defines the number of total mini sources. Assuming the average ‘off’ frequency $\lambda$ to be 0.4 /sec, the simulation results in Fig. 3 depict curves relationship between the original function (1) and the simplified function (7). Keeping $N = 60$ and fixing the probability of buffer overflow to $10^{-1}$, minimum buffer size should be equal to 48, through equation (1), while it should be 45 through equation (7), and keeping $N = 55$ and fixing the probability of buffer overflow to $10^{-2}$, minimum buffer size should be 50 through equation (1), while it should be 52 through equation (7). These results show that the differences are negligible and the simplified equation is acceptable.

V. CONCLUSION

In this paper we have provided a simplified formula for fluid model to investigate the probability of buffer overflow $G(x)$ and buffer size $x$ in wired backbone PAN. We conclude that the simplified formula provides acceptable results for the implementation of fluid model. However, simple equation can be utilized in future for the further wireless PAN.

REFERENCE


Detection of Signals in Rayleigh Fading Channel with MUSIC Algorithm

Syed A. Raza, Noman Shabbir, and Linsheng Li

Abstract—This paper concerns on the effects of Rayleigh fading channel on the frequency estimation of sinusoidal signals with multiple signal classification (MUSIC) algorithm. We find out the relationship between the maximum Doppler shift and the frequency estimation offset in frequency-flat Rayleigh fading channel. The proposed method largely reduces the calculation time when dealing with multiple signals as it shares the algorithm with the direction-of-arrival (DOA) estimation which can also be done by MUSIC.

Index Terms— Doppler shift, MUSIC, Rayleigh fading channel

I. INTRODUCTION

The processing of multiple signals has become quite important in smart antennas and wireless communication. For instance, multiple signal classification (MUSIC) is implemented to provide estimates of the number of signals, direction of arrivals and polarization [1], [2]. Also the carrier synchronization is very important and this is often performed by phase differential algorithm or Fast Fourier Transform (FFT). However, these two algorithms consume a lot of time in dealing with multiple signals. Here we propose MUSIC to estimation of carrier frequency detection.

Rayleigh fading can be a useful model in heavily built-up cities where there is no line of sight between the transmitter and receiver and many buildings and other objects attenuate, scatter reflect, refract and diffract the signal.

Our motivation is to find out the exact relationship between the parameters of the Rayleigh fading channel and the frequency offset which is the difference of the estimation frequency and the original one. Here the frequency-flat Rayleigh fading channel is studied and we only consider one parameter for this channel: the maximum Doppler shift. And a sinusoidal signal is used as the carrier. And from the simulation result we can see that MUSIC is suitable to detect sinusoidal signals in frequency-flat channel.

II. REVIEW OF THE STATE OF ART

There are two types of algorithms that can be used to detect sinusoidal signals in Rayleigh fading channel: phase differential and FFT [3]. These two methods provide Doppler compensation but they consume too much time and have a high complexity in calculation when dealing with multi signals. So we have to find out some more efficient algorithms to deal with multiple signals.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Our research question is how we can distinguish different signals after passing through a Rayleigh fading channel and what the impact on the signal is if we change the Signal to Noise Ratio (SNR) and the Doppler shift of the signal.

The main problem is that we have a Rayleigh fading channel. As there are multiple copies of the same signal at the receiver so we can distinguish them by using the MUSIC algorithm. It returns the pseudospectrum of the signal and we can identify different signals with help of it.

We use the MUSIC algorithm to detect sinusoidal signals in Rayleigh fading channel and find out the exact relationship between the Doppler shift of the channel and the detection frequency offset under certain SNR level.

IV. PROBLEM SOLUTION

A. Simulation Model

In order to test whether MUSIC can be used to detect sinusoidal frequency signals in Rayleigh fading channel and if it can we will have to find out the deviation between the original signal frequency and the detection one. We do this simulation as shown in Fig. 1. We set the parameters such as maximum Doppler shift and SNR in Rayleigh fading channel and then use MUSIC algorithm to get the spectrum of the signal. After this we compare the detection result with the original frequency and get the offset between these two.

We do the whole simulation in MATLAB and the MUSIC algorithm is implemented as a function named PMUSIC. This function returns the pseudospectrum of the input signal.

In one of our simulation examples we are adding three different sinusoidal signals with frequencies which are given by the user in MHz As the signal is transmitted and in the communication channel Rayleigh Fading is added to the transmitted signal and along with Rayleigh Fading the Additive White Gaussian Noise (AWGN) is also added to make the scenario more practical. The Rayleigh Fading is dependent on

Fig. 1 Simulation model
two factors that are the sampling time and the Doppler Shift frequency. These two factors can be defined as:

\[ F_d = \frac{v}{\lambda} \]

\[ T_s = \frac{1}{f_c} \]

Where

\[ F_d \]: Doppler Shift frequency  \\
\[ f_c \]: Carrier Frequency  \\
\[ c \]: Velocity of Light  \\
\[ v \]: Speed of the Mobile User

In the simulation first we are taking the carrier frequencies of 45, 46 and 47 MHz respectively and then add them. The speed of the moving object is 60 km/hour and the SNR is 2 dB. After passing this input signal through Rayleigh Fading channel we apply the PMUSIC function on this input signal and the signal after the Rayleigh Fading to detect their frequencies as shown in Fig. 2

**B. Simulation results**

In Fig. 2 we can observe the frequency difference between input and the output signals. To find the exact relations between the Doppler Shift frequency, frequency offset and SNR, we take different values of Doppler Shift frequency while SNR kept constant to find the relation between the Doppler Shift and frequency offset. Then in other two cases SNR is still constant but with different values. After that simulation we get the relation between SNR, Doppler Shift frequency and frequency offset as shown in Fig. 3.

**V. CONCLUSION**

In this paper we use the MUSIC algorithm to detect sinusoidal signals in Rayleigh fading channel and from the simulation results we can find out that MUSIC suitable to do this. In Fig. 3 we can find out the exact relationship between SNR, maximum Doppler shift and detection frequency offset.

From the simulation results we can see that if the maximum Doppler shift is bigger than 0.025F_s the detection frequency offset will increase very fast and so in this situation the detection results will have a big deviation from the actual one. Also we can see that the detection frequency offset will become bigger if we decrease SNR when Doppler shift is smaller than 0.025F_s but opposite result will occur when Doppler shift is bigger than 0.025F_s. This means that increasing SNR will not improve the detection result but make it even worse when Doppler shift is bigger than a certain value. An equalizer is necessary in this situation and it is the future work.

In future we will compare MUSIC with phase differential and FFT methods to show that MUSIC is more suitable to deal with multi signals because it saves computation time and decrease the calculation complexity. Also we will implement some methods which are used to improve the performance of MUSIC in calculation complexity and so on[4].

**ACKNOWLEDGMENT**

We thank Prof. Wlodek a lot for his help in our project and his helpful suggestions.

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An Improved LMS Adaptive Filter for Noise Cancellation in ECG Signal

Dandey Venkat Prasad, MD. Anamul Hoque and Abid Aziz Khan

Abstract—Noise cancellation in ECG signal using Adaptive filter is proposed. The mean squared error between the ECG signal which is noisy and a reference input is minimized using adaptive filter. An Adaptive filter structure is introduced mainly to cancel the intrusion 60 Hz power line interference and baseline wander noise. It works on Least Mean Square (LMS) technique to eliminate the noise that is correlated in some way with the ECG in the primary input. The selection of better convergence parameter and proficient use of discrete time reference signal improved the LMS technique and eliminated the noise in ECG signal.

Index Terms— Noise cancellation, ECG, Adaptive Filter.

I. INTRODUCTION

THE Electrocardiogram (ECG) recording has an indispensable role in cardiology. Idyllic, Simple, effective and low cost analysis of ECG will have more impact on patient life and also on social costs. Many microprocessor based recorders are invented to trail on time signal processing. Due to the constraints of computational power of microprocessor there is need of digital filters to purge the noise. In [2], N. V. Thakor and D. Moreau have developed integer coefficient and quantized coefficient digital filter respectively for real time execution by microprocessors for ECG analysis. The literature in this topic introduces an improved adaptive filtering technique (Least Mean Square) to purge the noise from ECG signal which is simple and effective in mathematical computations.

II. REVIEW OF STATE OF ART

In [1], to cancel the 60Hz power line noise, motion artifact and baseline wander noise different filter structures are presented. The fundamental idea about Adaptive filtering has been given in brief by Widrow. Many of the ECG processing applications are based on this idea. The idea in [3] is - A reference signal which is power line interference from area not in the vicinity of the ECG recording is used to cancel the interference of power line from the ECG.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Cardiac rhythm, cardiac ischemia and ventricular hypertrophy are the main disturbances in ECG which is due to interference of signals from other sources in the vicinity of the patient. Equipment to analyze and cancel noises more than one at a time is required. Amalgamation of many factors proclaim for research in electrocardiography. They are:

1. Diagnosis of ECG using low-cost high performance technology.
2. Design of portable devices and making the patient less hospital dependent.

To develop portable devices and have human computer interaction with microprocessor, there is requirement of less computational algorithm techniques. In this paper improved LMS adaptive filtering technique is applied to cancel the 60 Hz power line interference and baseline wander noise. This is done by selecting better convergence parameter and proficient use of reference signal in discrete time. Simulation is done in MATLAB and the results shows fast convergence and purge of noise.

IV. PROBLEM SOLUTION

System Model:

The primary input to the filter is the ECG signal \( x(k) \) with additive noise and \( u(k) \) is the reference signal which is noise. The error between the initial signal and the product of manipulated version of \( u(k) \) assigned as \( \text{reg} \) and updated weight \( w \). The error equation is shown below:

\[
e(k+1) = x(k) - w \cdot \text{reg}.
\]

Minimizing the Mean Square Error results in a filter output that is the best least-squares estimate of signal \( x(k) \).

The LMS algorithm is an iterative technique for minimizing the MSE between the primary and the reference inputs. The LMS algorithm is used to obtain the filter coefficients or weights given by.

\[
w(k+1) = w(k) + \beta \cdot \text{reg}.
\]

Where \( w(k) \) is set of filter weights at time \( k \), \( \beta \) is selected randomly to produce convergence at desired rate.

Time constant for convergence is \( 1/(4\beta \alpha) \).

For stability the parameter \( \beta \) is chosen between \( 1/\alpha \) and 0. \( \alpha \) is the largest eigenvalue of the auto correlation matrix of the reference signal.

V. FILTER IMPLEMENTATION IN MATLAB

The main task is cancellation of the diverse noise contributions to ambulatory ECG. The topic in next page reviews the design and application considered in this paper.
Baseline Wander and 60 Hz power line noise Reduction

The improved LMS algorithm based adaptive filter is simulated in MATLAB. The predefined ECG signal is taken which is given by function \( \text{ecg}(500) \) which is of 500 Hz frequency.

\( x(k) \) is the signal which is added ECG signal with noise \( u(k) \). Again \( u(k) \) is taken as reference signal to find error. Weight of the filter is update as shown in (2). Error is found as shown in (1). \( \beta \) is selected as 0.005 randomly and changing it between \( 1/\alpha \) and 0. The flowchart for the iteration process is shown below.

\[
\begin{align*}
\text{Start} & \quad N = 1000 \\
\text{Reference noise signal} \quad \text{Input signal (ECG with Noise)} \quad \text{Display i/p signal} \\
\text{Manipulate the shifted signal with flipud command} \\
\text{Update weight vector} \rightarrow \text{Error b/w } x(k) \text{ and noise} \\
\text{N } & \leq 1000 \\
\text{End} & \quad \text{Yes} \quad \text{No}
\end{align*}
\]

Fig. 1. Data flow diagram for LMS algorithm

Filter length of \( N=1000 \) is initialized. Noise signal is taken as reference and also to mix with ECG signal. The simulated ECG signal is mixed with noise for our analysis. Then reference signal is shifted and weight vector is updated. Finally error is found and the above steps are repeated until \( N \). The error signal is the noise less ECG signal.

VI. RESULTS

Simulated ECG signal of 500Hz is shown in Fig.2.

![Simulated ECG signal of 500 Hz](image)

Fig. 2. Simulated ECG signal of 500 Hz

(a)

![Image of graph](image)

(b)

Fig. 3. (a) Noisy ECG signal. (b) ECG signal after filtering

Fig. 3(b) shows the ECG signal added with Baseline Wander Noise and 60 Hz power line interference noise. Due to better selection of \( \beta \) and proficient use of discrete time noise signal has given deviations for a short time and it reached convergence 50% faster (compared to normal LMS Adaptive Filter). The Fig. 3(c) shows the recovery of simulated ECG signal which is noise free and performance of the filter.

VII. CONCLUSION

The achievement of this paper is purge of base line wander noise and 60Hz power line interference using improved LMS algorithm based Adaptive Filter. The important feature of this filter is, without past information the primary signal is predictable and is easy to implement on modern microprocessors with numeric abilities.

In ambulatory ECG signal many forms of noise occur unpredictably. So, there is a need of multistage filter structure to eliminate the all forms of noise. There is also a need of Artificial Neural Network tools for perspective categories that are able to deal even with high sensitivity to noise and ambiguous patterns.

REFERENCES


Trade off Between VoIP Security and QoS: SRTP Based Approach

Ayanda Hafeez Olabanji, Gillani Syed Fakhar Uz Zaman, Fahad Razzaq

Abstract—Commercial deployment of voice over internet protocol has gathered momentum in recent years. A major issue with the VoIP network is in improving QoS and security. This paper discusses the solution and challenges involved in providing QoS for voice traffic. Also, we focus on the QoS for the secure media session using Secured Real Time Protocol. This paper investigates that securing voice packets can ensure the quality of service. A simulation is conducted using OPNET modeler 14.5 for VoIP traffic using both RTP and SRTP. The results of the performance analysis show the impact of security on VoIP QoS.

Index Terms—OPNET, RTP, SRTP, Security and VoIP.

I. INTRODUCTION

Voice over IP is the exchange of voice packets over internet and is for sure the cheapest way of communication in the present era. The traditional Public Switched Telephone Networks will soon be overwhelmed by VoIP services as it provides variety of different services under an umbrella, but there exist some apprehensions which must be dealt with.

QoS and security are the factors which are of main concern while comparing VoIP to traditional PSTN since VoIP doesn’t guarantee the same QoS as well as security which are being offered by PSTN.

II. REVIEW OF THE STATE OF ART

There exists a tradeoff between QoS and Security issues while talking about VoIP. Many different protocols exist which are used to protect VoIP traffic from different types of attacks such as man-in-middle-attacks, eavesdropping, Denial of Service, Spam over Internet Telephony and many more. Previously VoIP was secured by IPsec and TLS but these protocols made an increased overhead for voice packets.

VoIP conversation is said to be secure when signaling and media sessions are well protected from attackers. These both can be secured by variety of protocols such as:

SIP: Session Initiation Protocol is responsible for the signaling session, once the connection is established media session takes over the control. While talking about security in SIP based networks it’s not an easy task [4]. SIP itself needs other protocols to ensure security like IPsec or TLS.

SRTP: Secure Real time Transport Protocol is a profile of RTP which can be used for secure transmission of data between two or more communicating entities. It provides message authentication, confidentiality, replay protection and integrity to media traffic over internet. SRTP obtains the keying material through secure hash function via single master key [4]. We can have different protocols for exchange of keys in SRTP which includes MIKEY, Diffie-Hellman algorithm, and PKI.

![SRTP process model](image)

Fig.1. SRTP process model

ZRTP: Another extension of RTP which is known as ZRTP [1]. It uses Diffie-Hellman algorithm for key exchange whenever a call is placed, so this does not need a third party for securely exchanging the keys. To overcome man-in-middle attacks, ZRTP uses Short Authentication String (SAS) which is mainly a cryptographic hash of two Diffie-Hellman values.

ZRTP is not yet a standard RFC but it is in the form of a draft in IETF and has been implemented in Zfone [5].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

QoS and security are the most important factors which must be taken care of in VoIP networks. The aim of the paper is to improve these two factors, as these both depend on each other critically. End-to-end delay (most critical), packet loss and Packet Delay Variation are the parameters which must be taken in consideration while securing end to end VoIP communication. In this paper, we analyzed the effect of security on VoIP traffic and evaluated the solution, which results in enhanced security and tolerable QoS.

We implemented the modeled network using OPNET modeler 14.5 [3].

IV. PROBLEM SOLUTION

Initially, VoIP traffic is implemented with G.711 speech codec and using RTP layer to transport the media stream. The G.711 speech codec was chosen because it helps to take into account the worst case in bandwidth, and get the best in quality. Figure 2 shows the network topology. The network consists of IP router, Switches, VoIP Gateway, HTTP server,
Cache Proxy, Email Server, Application Definition, Profile Definition, and Work Stations. The link speed to every node is 100mb/s. There are 3 computers and 1 switch at each subnet. We used the predefined voice application and each workstation in the subnet is configured to implement the application.

SRTP Process Model: The first scenario of the model is implemented with the RTP protocol. In modeling of the SRTP protocol, both AES (Advanced Encryption Standard) and Diffie-Hellman algorithm are implemented to secure the exchange of encryption keys.

![Fig. 2. Network Topology Model](image)

Simulation results: OPNET was configured to get graphed results for VoIP traffic. The duration for the simulation run was set to 25 minutes. The background traffic is generated, which is by default 40 seconds in OPNET starting from the beginning of the simulation run.

Graph of end to end delay of encrypted VoIP packets between nodes, Jitter, and Packet Delay variation are displayed. It is noted that the consequences of delay are some of the most crucial aspects affecting VoIP service.

![Fig. 3. End to End delay](image)

While examining the figure above, it is observed that the difference in end-to-end delay between VoIP traffic and SRTP VoIP traffic is less than 150ms throughout. This was highlighted when the graph was zoomed out. However, one-way delay of 150ms is acceptable for voice application.

![Fig. 4. Jitter](image)

For voice communication the maximum acceptable jitter is 50ms [2]. As shown in Fig. 4, the maximum value of jitter is far less than 50ms for both VoIP and Secured VoIP traffics, which is an appreciable result.

![Fig. 5. Voice packet delay variation](image)

It is observed that the highest value of packet delay variation reaches 5.6sec for the secured streams, while it was 4.8sec for the unsecured stream. With this, we are able to confirm that the use of SRTP protocol have effect on the quality of service of VoIP. However, the increase in delay of the end-to-end packet stays lower than 150ms, which is the most important parameter, as prescribed in ITU G.1010 recommendation.

V. CONCLUSION

We successfully modeled with both RTP and SRTP streams. Based on the modeling results, we observed that at every instance the delay of voice packet did not exceed the 150ms value. From this, we can assurely propose that quality of service is ensured while securing voice packets by SRTP.

In the future we plan to extend this research by investigating VoIP QoS using ZRTP, and then compare it with SRTP to ensure better QoS and improved security.

REFERENCES

Simulation of QPSK and 16-QAM for Estimating BER and SER through Rayleigh Fading Channel

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Abstract: This paper reveals the comparison of performance of Bit Error Rate (BER) and Symbol Error Rate (SER) using mathematical model and simulation for the two modulation techniques, Quadrature Phase Shift Keying (QPSK) and 16-Quaternary Amplitude Modulation (16-QAM) with an Additive White Gaussian noise (AWGN) through Rayleigh Fading Channel.

Index Terms—Bit Error Rate (BER), Rayleigh Fading Channel, QPSK, 16-QAM.

I. INTRODUCTION

In telecommunication field the major challenges is to convey the information as efficiently as possible through limited spectral width, though the information is lost in most of the cases and signal which is sent originally will face fading. To reduce the bit error rate the loss of information and signal fading should be minimized.

In our paper we analyze two modulation techniques, QPSK and 16-QAM to reduce the error performance of the signal and compare which technique is better through Rayleigh Fading Channel in the presence of AWGN.

II. REVIEW OF THE STATE OF ART

After reviewing several latest references, we find that the way to compare QPSK and 16-QAM through the Rayleigh fading channel, using the constellations and also the BER for the modulation techniques. In [4], it uses signal-space concepts to efficiently evaluate the performance of M-PSK and M-QAM that is considering the orthogonal axis and then observing which point is belonging to which phase and then coding it. In [2], it calculates the BER and peak to average power ratio (PAPR) to analysis the performance. The main contribution of the research project is, we add the Rayleigh fading channel which makes the condition tough and more realistic to test the modulation techniques and we are using matlab simulation to validate the mathematical calculations.

IV. SYSTEM MODEL

We state a mathematical model which is shown in Fig.1.

A. Rayleigh Fading

The Envelope of the sum of two quadrature Gaussian noise signals obeys Rayleigh distribution.

\[
Rayleigh\ Fading = \frac{\sqrt{x^2+y^2}}{2}
\]

B. QPSK Performance of Bit Error Rate

The Bit Error performance for the various digital modulation systems is directly related to the distance between the points on a signal space diagram. We can see it from Fig. 2.

The bit error probability of a QPSK system is

\[
P_e = \frac{1}{\log_2 M} \text{erf}(Z)
\]

Where:

\[
\text{erf} = \text{error function}
\]

\[
Z = \sin\left(\frac{\pi}{m}\right)\sqrt{\log_2 M} \left(\frac{I_m}{N_0}\right)
\]

AWGN

Signal \rightarrow QPSK \rightarrow Rayleigh Fading \rightarrow output

III. RESEARCH QUESTION AND PROBLEM STATEMENT

How to estimate the difference in BER&SER for two modulation techniques, QPSK and 16-QAM, in Rayleigh fading channel under AWGN? Here we will use signal to noise ratio (SNR) for mathematical calculation which is used to validate the simulation.

The main contribution of the research project is, we add the Rayleigh fading channel which makes the condition tough and more realistic to test the modulation techniques and we are using matlab simulation to validate the mathematical calculations.

Fig.1. Model of the Problem.
C. QAM error performance

The bit error probability of an L-level QAM system is

\[ P(e) = \frac{1}{(\log_2 L)(L-1)} \text{erfc}(Z) \]  

\[ Z = \sqrt{\frac{\log_2 L}{L-1}} \frac{E_b}{N_0} \]

Where:
\[ \text{erfc} \] is the complementary error function.

The minimum bandwidth required to pass the output of a 16-QAM modulator is equal to one half that of the QPSK.

![QPSK and 16-QAM constellations](image)

Fig.2. (a) QPSK constellation with Gray coding (b) 16-QAM constellation with Gray coding.

V. SIMULATION AND RESULTS

A simulation work is done based on the system model described previously. In the simulation part, the performance of QPSK and 16-QAM is measured in AWGN environment through multipath Rayleigh fading channel.

In our simulation, 1000000 random bits are generated to detect channel performance in multipath Rayleigh fading in AWGN environment, using the Gray coding to define the QPSK symbols and 16-QAM symbols. Raised cosine filter is used to emulate transmission medium and SNR is varied from 0 to 31 dB depicted in Fig.3. In the simulation work, first we compared the performance in AWGN environment to know which technique is better for not so tough condition, and then we added multipath Rayleigh fading, this time the condition is even worse than before. Comparing the theoretical values which are calculated in the system model with the simulation values of BER depicted in Fig.3. We can see that they are almost insuscibled. From Fig.3 we also can see that BER/SER is smaller in QPSK than in 16-QAM at the same value of SNR. If we want to get the same BER of QPSK and 16-QAM, we should increase the transmitted power, from the Fig.3.b), we can see that it almost has to increase 30dB.

But on the other hand, each symbol of QPSK convey 2 bits but that of 16-QAM is 4 bits/symbol, so in time domain equivalent symbol period of 16-QAM is twice as long.[2] This means that when we transmit the signal, the 16-QAM has better bandwidth efficiency.

![Comparison of performance](image)

Fig.3. Comparison of performance (a) of QPSK and (b) of 16-QAM under Rayleigh fading and AWGN.

VI. CONCLUSION

In this paper, a comparison has been performed between symbol error rate (SER), bit error rate (BER) and also the theoretical BER for QPSK and 16-QAM. First we derived the BER formula for the two modulation techniques, and then the matlab simulation was managed to verify the accuracy of our theoretical results. Finally it could be concluded that BER performance of QPSK is better than that of 16-QAM at the expense of spectral width. The comparison of performance in AWGN and Rayleigh fading channel indicates a significant fact that QPSK modulation is more suitable for tough condition. In our paper, we use BER to calculate the errors of per bit, later in the future, we will add coding in the system model, so SER will depend on the coding type, and also in the future, we will concentrate on performance evaluation of multiuser systems on more realistic fading channel model.

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Speech Recognition Using Hybrid Algorithm

Nagarajesh Garapati, Kalyan Chakravarthy Talluri, and Zhao Jie.

Abstract—Speech recognition using neural networks is familiar now a day. The main aim of this paper is to minimize the training time of Neural Networks (NN) in speech recognition by using Steepest Descent Method (SDM) and Improved Genetic Algorithm (IGA) together. Here we obtain results by training neural networks by SDM then we train these results by NN for better performance.

Index Terms—Neural networks, steepest descent method, improved genetic algorithm, speech recognition, Hybrid algorithm.

I. INTRODUCTION

SPEECH recognition is very popular in the field of voice detection. In recent years, Improved Genetic Algorithm (IGA) has been employed in the research of speech recognition to train Neural Networks (NN) because of its excellent learning performance. However IGA has its own disadvantage such as time consuming procedures.

In this paper, we are attempting to minimize the training time of NN in speech recognition by using Steepest Descent Method (SDM) and IGA.

II. REVIEW OF THE STATE OF ART

The traditional way to train the NN for speech recognition is by using SDM. After that SDM was replaced by Genetic Algorithm (GA) to improve the results up to 90% [1]. Now IGA came into picture to work even more efficiently with almost 95% accuracy [3]. IGA will take more time to train the NN. So we are trying to overcome that time consuming process by replacing the initial guess of the NN with the results of SDM in speech recognition.

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Our question is mainly focused on how to reduce training time of the NN in speech recognition. Several steps are involved in IGA, to solve the problem of time consuming process, we found two methods: the first method is to change the whole IGA process to Dynamic Time Warping Model or Hidden Markov Model. However the two models referred cannot be used here because of low satisfaction of accuracy. The second method is by using the result of SDM and inputting the result to IGA to realize speech recognition. The second one is a new idea to solve these problems.

IV. PROBLEM SOLUTION

The NN trained by SDM or IGA are used on prediction of nonlinear systems for speech recognition. For speech recognition we considered sampling frequency of 8 kHz, mono channel 16 bit sampling point. After that point detection followed by pre emphasis was made. With the combination of SDM and IGA, we present a new algorithm to achieve performance of the identification of nonlinear systems. The Hybrid Algorithm (HA) part is described in the Fig. 1.

![Fig. 1. Two stages of Hybrid Algorithm [1].](image)

The SDM is used to train the weight and also to train the bias of NN for a long time, it is easy to reach a local optimum, and recently the IGA is used to replace the SDM on the training of NN because it reaches the global optimum. So the learning performance of SDM and IGA are complement for each other.

In actual IGA process, it will start to find the global optimum by choosing some initial weights, these weights might be so far away from the solution, so IGA needs to perform quite a lot of iterations to reach global optimum, and takes longer time for training [2].

So we have chosen the initial guess which is very near to solution, for that we used SDM result weights. Now we replace these weights to the initial weights of the IGA. Then IGA starts its iterations with SDM results, and it will take less time to come up to global optimum.

Fig. 2 illustrates HA that combines both SDM and IGA showed in it. In the first step SDM chooses random weights to start the process. In the process we suppose the system is nonlinear one, and the orthogonalized SDM is an efficient algorithm which not only can overcome drawbacks of a positive definite matrix but also can exploit the convergence speed of the matrix. Here we employ the formula in SDM:

\[ F(x) = KP[Sx]b \]  

(1)

Where the coefficient matrix \( K \) is \( n \times m \), the coefficient matrix \( S \) is \( m \times n \), the map \( P:R^m \rightarrow R^n \), which is a nonlinear matrix, \( b \) belongs to \( R^m \) and \( x \) belongs to \( R^n \) are known and unknown vectors.

The NN for speech recognition is been trained by SDM which continues until Mean Square Error (MSE) reaches 10%,
now here we will use the result as the input to IGA and then, IGA starts execution towards to global optimum

The main IGA consists of,

\[ \text{osc}_1 = [\text{os}_1^1, \text{os}_2^1, \ldots] = (P_1 + P_2)/2 \]  

\[ \text{osc}_2 = [\text{os}_1^2, \text{os}_2^2, \ldots] = P_{\text{max}}(1-W) + \max(P_1, P_2) \]  

\[ \text{osc}_3 = [\text{os}_1^3, \text{os}_2^3, \ldots] = P_{\text{min}}(1-W) + \min(P_1, P_2)W \]  

\[ \text{osc}_4 = [\text{os}_1^4, \text{os}_2^4, \ldots] = ((P_{\text{min}} + P_{\text{max}})(1-W)(P_1 + P_2))/2 \]  

Where \(\text{osc}_1^1 \sim \text{osc}_4^4\) are chromosomes of next generation \(P_1, P_2\) are the two chromosomes chosen from the parent, and the parent data comes from the results of the SDM. Max (P1, P2) and min (P1, P2) are the new chromosomes, in which each gene are the maximum and minimum, respectively, of the genes in the two chromosomes P1 and P2.

Here are the results from the previous algorithms and the HA are described above. From these results you can observe clearly both accuracy (in terms of MSE) and the time required for training the NN are achieved. By observing the graphs in Fig. 3 and Fig. 4 for IGA and HA, we can judge that good efficiency of speech recognition can be achieved in less time.

In Fig. 3, IGA achieved the output by taking very long time more than 150 minutes to come to a satisfied value. In Fig. 4, the HA achieved a good efficiency and it consumed less time than IGA approximately 100 minutes to train the NN.
Abstract—Bit Error Rate (BER) is an important term in any communication system. In this paper, the effects of increasing the Signal to Noise Ratio (SNR) on BER in the presence of Additive White Gaussian Noise (AWGN) with and without Orthogonal Frequency Division Multiplexing (OFDM) using the Binary Phase Shift Keying (BPSK) as the modulation scheme is investigated. Since, there are several modulation techniques are already to use reducing BER, it has becomes the choice of time to choose the right one which is simple and has the ability to handle BER effectively. Our studying has shown that OFDM is the best technique for reducing BER.

Index Terms—BER, BPSK, OFDM

I. INTRODUCTION

Orthogonal Frequency Division Multiplexing (OFDM) has become a popular transmission technique for high data rate wireless communication in recent years. OFDM is a spectrally efficient form of frequency division multiplexing that divides its allocated channel spectrum into several parallel sub-channels.

In OFDM, each user is allocated several carriers transmitting their data. The transmission is generated in such a way that all carriers are orthogonal to each other. This allows them to be packed together much closer and make the transmission more spectral efficient, robust against Inter-Symbol Interference (ISI) and fading caused by multi path propagation.

Bit error rate (BER) increases in the channel when the data rate is increased. The BER cannot be eliminated completely but can be minimized using OFDM.

In future, OFDM use in wireless and broadband system applications like digital TV and audio. The main advantage of OFDM is single carrier scheme means attenuation of high frequencies in a long copper wire, narrowband interference and frequency-selective fading due to multi-path. OFDM is used in WiMAX and in future OFDM is Fourth-Generation (4G) technology.

II. REVIEW OF THE STATE OF ART

The addition of AWGN in the transmitted signal produces the critical effect on the received signal in terms of frequency, phase and time offset. [1]

In BPSK modulation with OFDM as a multiplexing technique the Co-Channel Interference and Noise has been significantly reduced and thus it also effects the BER. [2]

III. RESEARCH QUESTION AND PROBLEM STATEMENT

The aim of this paper is to investigate the effects of increase the data rate on SNR in BPSK; while OFDM is applied to reduce the BER.

IV. PROBLEM SOLUTION

A. The proposal method

In this project is to find a way to reduce BER in BPSK modulation and to analyze the performance of the BER using MATLAB-ver2007 for simulation and this simulation generate two different graphs, one for BPSK modulation and demodulation without OFDM and second graphs for BPSK modulation and demodulation with OFDM.

B. Simulation Model

The first we modulation and demodulation of BPSK without increasing data bits rate, BER is compared and no errors were found. No channels are increase because no bits are increase. BPSK can not handle bit error rate by using other modulation techniques. In Fig. 1 the block diagram shows how to modulate and demodulate the data rate in BPSK.

![Fig. 1. Block diagram](image)

C. Simulation Scenario-1

In Fig. 2 the bit rate are increased and, BPSK modulation and demodulation is done again. This process is shown in Fig. 2. Cosine wave is used with increased bit rate to simulate
BPSK. After performing comparison between the transmitted bits and the received bits; more bit errors (blue lines) were found. Because when the bits are increased then ultimately channel also is increased, channel is directly proportional to bits and BER depends upon error per number of bits. For solving this some modulation technique are required.

Fig. 2. BER increased

D. Simulation Scenario-2
For reducing the BER we are adding OFDM modulation technique. In Fig. 3 the same process is repeated with OFDM modulation and we may observed the BER is reduced. This process is shown in Fig. 3.

The error reduces on the simulation-based results represent the error are reduced on the results obtained for the number of bits transmitted.

Fig. 3. BER reduced

V. CONCLUSION
In this paper, Bit error rate of the transmission system with and without OFDM scheme is compared and results are obtained through MATLAB simulation. This Paper shows that bit error rate, using BPSK modulation, can be reduced through OFDM technique.

OFDM is used in wireless and broadband system applications like digital TV and audio. OFDM is also used in WiMAX and in future OFDM we predict Fourth-Generation (4G) technology.

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REFERENCES
Performance Improvements of IP Networks using MPLS: Throughput, Delay and Jitter Comparisons.

Bained Nyirenda, Ahsan Haroon, and Imran Ikram

Abstract—In this paper we present a comparison of throughput, delay and jitter of Open Shortest Path First (OSPF) and Multi Protocol Label Switching (MPLS) based networks. MPLS is an effective approach in dealing with congestion in IP networks and thus improving their performance. MPLS guarantees Quality of Service (QoS) of IP networks. Many network carriers are facing the problem of how to accommodate ever-growing demands for bandwidth. Therefore, MPLS traffic engineering is proposed as a technique to help mitigate this constraint. By taking advantage of MPLS, traffic engineering (TE) can route the packets through explicit paths to optimize network resource utilization and traffic performance.

Index Terms—MPLS, OSPF, TE, QoS.

I. INTRODUCTION

The proliferation of bandwidth intensive and QoS aware applications has placed an enormous challenge on the Internet infrastructure. Most Internet protocols fail to efficiently handle this traffic demand due to their inability to effectively engineer traffic as they tend to place routing emphasis on least cost paths. MPLS was developed to offer added features in order to address this deficiency. Our work is motivated in determining the extent to which basic MPLS improves the performance of an IP network in terms of throughput, packet end-to-end delay and jitter.

This paper analyses the throughput, delay and jitter performance of pure IP networks and MPLS enabled networks. On the basis of these parameters a comparative analysis of an OSPF network and an MPLS network is done. Our results indicate that basic MPLS offers considerable improvements in the performance of IP networks.

II. REVIEW OF THE STATE OF ART

OSPF is a broadly used gateway protocol which can accommodate rapid changes in networks [1]. It uses open shortest path first algorithm to compute paths between a source and destination. However, OSPF fails to adequately distribute traffic amongst routes of unequal costs and thus causes overutilization of some paths whilst others are underutilized. With small modifications, it can be used with MPLS based networks [1] to enhance the performance and scalability of backbone networks [2].

MPLS is an emerging technology which plays an important role in providing QoS and TE features to IP networks [2]. It is helpful in managing traffic when different paths in a network are either underutilized or over utilized [3].

MPLS is based on switching technology. Packets are switched through the network by swapping labels. An MPLS network consists of Label Switched Routers (LSRs) in its core and Label Edge Routers (LERs) at its edges. A Label Switching Path (LSP) is created between the LERs to provide a connection oriented path for traffic flow based on some QoS level [3]. Path setup requires a signaling protocol such as Resource Reservation Protocol (RSVP) or a Label Distribution Protocol [3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

Our study provides a comparative analysis of throughput, end to end packet delay and jitter of OSPF and MPLS networks. Throughput is the average rate at which the destination will receive packets. It is important for a network to have a high throughput for high performance. Packet delay is the time the packet spends in transit and jitter is the variation in the arrival rates of packets at the destination. For time sensitive applications such as voice, it is necessary for the network to provide low delay and packet jitter to maintain acceptable QoS levels.

A. Research Question:

In this paper we answer the question of whether basic MPLS improves the throughput, delay and jitter of an OSPF network in an environment of congestion.

B. Problem Statement:

MPLS greatly improves the performance of IP networks in terms of QoS parameters such as throughput, delay and jitter. Our contributions include:

- Ascertaining that a basic MPLS network without any advanced TE techniques gives a pronounced improvement in the throughput, delay and jitter of an OSPF network. Thus we provide a platform for network providers to use MPLS in their networks.
- We provide a basis for further research into MPLS based TE employing techniques such as DiffServ.
Performance simulations using OPNET Modeler 11.0

IV. PROBLEM SOLUTION

We observed the effects of UDP traffic on congestion sensitive TCP traffic by varying the amount of UDP traffic input into a simple network. Throughout the experiment, TCP traffic was held constant at 1 Mbps.

Firstly a simple OSPF network was simulated to provide baseline statistics. It was seen that in this scenario, all the data from the TCP and UDP sources was forced through the link with the least cost.

Secondly, an MPLS enabled network was modeled under the same traffic conditions. We enabled MPLS on the network with two Label Switching Paths (LSPs). We separated the UDP and TCP traffic into different Forwarding Equivalence Classes (FECs) and bounded each to a separate LSP. In this way, we were able to compare the results of the two scenarios under the same underlying conditions.

A. Simulation Setup

We set up the network as depicted Fig. 1. Two traffic sources were modeled; a TCP source and a UDP source. There were two parallel paths between the source and destination with a 1 Mbps link through LSR 3 (depicted by Red) and a 3 Mbps link through LSR 2 (depicted by Blue) in Fig. 1.

B. Results Analysis

The results obtained from the simulations are depicted in Fig. 2 and Fig. 3. When the aggregate traffic from the two sources was less than the link capacity (less than 3 Mbps), we observed that the TCP throughput for both the simple OSPF and MPLS networks was the same as shown in Fig. 2.

When more UDP traffic was sent over the link such that the aggregate traffic exceeded the least cost link capacity, the congestion sensitive TCP throughput was affected in the simple OSPF network. When TCP detects congestion on a link, it reduces its traffic input thereby allowing the UDP to send more traffic. This creates a vicious circle and the TCP throughput tends to be drastically reduced due to congestion despite the other route carrying no traffic. This situation was changed when MPLS was enabled on the network as the TCP traffic was engineered to take the underutilized path. This is also depicted in Fig. 2.

Fig. 3 shows the packet end-to-end delay and jitter respectively. It can be seen that packet jitter and delay are greatly reduced in the network with MPLS enabled. It can be concluded that, in our given conditions, MPLS had the effect of improving the TCP throughput and at the same time reducing the packet end-to-end delay and jitter. This is because it was possible to engineer the traffic onto least congested paths in the MPLS network. This is however not possible with IGP routing protocols such as OSPF which tend to concentrate traffic onto least cost paths.

V. CONCLUSION

This paper has shown that basic MPLS improves the performance of IP networks by enabling traffic to be engineered onto paths that are underutilized. This was studied using OSPF and MPLS networks.

It can also be concluded that MPLS greatly reduces the packet end-to-end delay and jitter as shown in this paper.

The paper has concentrated on basic MPLS and it would be interesting to quantify the improvements that can be achieved through the application of advanced TE techniques such as DiffServ, over MPLS.

ACKNOWLEDGEMENTS

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REFERENCES

Abstract—To collect correct data that is passed through multipath fading channel is a challenging issue in Orthogonal Frequency Division Multiplexing (OFDM). In this paper an attempt is made to estimate Channel Impulse Response (CIR) for OFDM signal over multipath fading channel using Least Square (LS) estimation algorithm. Channel estimation is done using comb-type pilot arrangement (CTPA). The technique used with comb-type pilot arrangement can efficiently estimate CIR with low error probability. We have investigated the performance of OFDM using LS channel estimation algorithm by measuring bit error rate over multipath fading channel with different modulation schemes.

Index Terms—Inter-symbol interference, Rayleigh Channel, Quadrature Amplitude Modulation (QAM).

I. INTRODUCTION

ORTHOGONAL frequency division multiplexing (OFDM) has emerged as one of the promising technique for data transmission over wireless channels in future generation networks [2]. It has high data rates and high spectral efficiency and is resistant to multipath channel fading effects [2]. In OFDM the spectrum of signal is divided into subcarriers which are orthogonal to each other. These are allowed to overlap, so spectrum can be used efficiently. Due to multipath effects different signals reach receiver with different delays so inter-symbol interference can occur between a signal and the signal following it. The use of cyclic prefix reduces the problem of inter-symbol interference (ISI).

Therefore an efficient channel estimation technique is required before demodulation of OFDM signal to get better estimates of the transmitted data since channel is time varying and frequency selective in mobile radio environment [1]. Least square channel estimation technique is used to estimate the channel before demodulation of OFDM signal [1]. This technique can efficiently estimate the channel with low probability of error. The LS technique is used under the condition that multipath delays are varying slowly and can be estimated by channel gain estimation. With the use of linearly modulated signal as pilots in sub channels, effect of multipath fading can be minimized.

II. REVIEW OF THE STATE OF ART

OFDM has been widely used in wireless communications due to its high bandwidth efficiency and high data rate transmission capability [2]. OFDM is an efficient scheme to minimize inter-symbol interference (ISI) over fading channels [1]. Least Square channel Estimation is used under the condition that channel is changed slowly [1]. The LS channel estimation technique can achieve low error probability by accurately estimating the Channel Impulse Response (CIR) and minimize errors due to rapid CIR variations [3].

III. RESEARCH QUESTION AND PROBLEM STATEMENT

As orthogonal frequency multiplexing is highly favorable scheme for transmission data over wireless channels, due to transmission impairments in multipath such as attenuation and inter-carrier interference the performance of OFDM is degraded.

Our research Question is to improve gain and reduce bit error rate for OFDM over multipath fading channel using least square channel estimation technique.

Our problem Statement is to find an efficient channel estimation technique to estimate the channel response to obtain correct data at the output. In this research we investigated LS channel estimation algorithm for OFDM signal transmitted over Rayleigh fading channel to obtain correct estimates at the output the scheme showed improvements in gain and reduction in bit error rate.

IV. PROBLEM SOLUTION

A. The proposed method

The baseband signal consisting of binary data and pilot signals are modulated using QPSK and M-ary QAM with M=4 and M=16. Pilots can be assigned by two types of arrangement. The first technique is Block type arrangement in which pilots are assigned to a particular OFDM block which is transmitted periodically. The second technique is comb-type pilot arrangement in which pilots are inserted uniformly in specific sub channels of OFDM block [3]. This type of arrangement gives us better estimation of CIR in rapidly varying fading channel. Cyclic prefix are inserted among data carriers to avoid inter-symbol interference. The resulting OFDM signal is transmitted over multipath fading channel.

After receiving the signal at receiver, guard intervals are removed. The guard intervals are longer than the length of CIR so that there is no interference between OFDM symbols. After this OFDM signal is demultiplexed. The received pilot signals \( Y_f(k) \) are extracted form received data. LS algorithm is applied on the received pilot signals and difference between received pilots and prior known pilots is calculated. The difference is minimized by using LS algorithm.
algorithm. The Channel Impulse Response is estimated for the received pilot signals [1].

\[ \hat{H}(k) = (E_pP_p(l_p)F_p^H)^{-1}(\sqrt{E_p}P(l_p)F_p^H)^H \]  

(1)

Where \( H \) is complex transpose, \( E_p \) is the energy per transmitted pilot symbol and \( l_p \) is location of pilots.

With this CIR, the transmitted data samples are recovered by dividing the received signal by the Channel Impulse Response.

\[ \hat{X}(k) = \frac{Y(k)}{\hat{H}(k)}, \quad k = 0, 1, \ldots, N-1. \]  

(2)

Simulation Scenario

The Channel Impulse Response for OFDM signal with different modulation schemes is estimated. The parameters used in simulation are given in TABLE 1. The Matlab tools are used to perform simulation. Guard intervals are kept longer than channel delay spread and behavior of channel is assumed to slowly varying. The pilot signals are uniformly distributed over OFDM symbol.

<table>
<thead>
<tr>
<th>TABLE 1</th>
<th>SIMULATION PARAMETERS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Symbol</td>
<td>Quantity</td>
</tr>
<tr>
<td>No of Subcarriers</td>
<td>256</td>
</tr>
<tr>
<td>No of Pilots</td>
<td>32</td>
</tr>
<tr>
<td>Guard Interval</td>
<td>64</td>
</tr>
<tr>
<td>Signal Constellation</td>
<td>QPSK, QAM, 16QAM</td>
</tr>
<tr>
<td>Channel Model</td>
<td>Rayleigh Fading</td>
</tr>
</tbody>
</table>

B. Simulation results

The simulation results when QPSK is used as modulation scheme are shown in Fig. 1.

The simulation results when M-ray QAM with \( M=4 \) and \( M=16 \) is used as modulation scheme are shown in Fig. 2.

The parameters used in both simulations are given in TABLE 1. The results showed that the system performance can be improved in terms of bit error rate by estimating channel response using LS algorithm with comb-type pilot arrangement. It is seen that for QPSK and QAM with \( M=4 \), BER is nearly the same at a given value of SNR. But for high data rate that is when higher order modulation is used, we need high SNR to achieve the same value of BER.

![Fig. 1. Signal to noise ratio v/s BER when QPSK is used as modulation scheme for OFDM over Rayleigh Fading Channel](image1)

![Fig. 2. Signal to Noise Ratio V/S BER when M-ray Modulation is used with M=4 and M=16 for OFDM over Rayleigh Fading Channel](image2)

The channel estimation is done under the condition that channel is slowly varying.

V. CONCLUSION

In this paper LS channel estimation method has been analyzed. We analyzed the bit error rate for OFDM with modulation schemes QPSK and M-ray QAM with \( M=4 \) and \( M=16 \) over Rayleigh fading channel using comb-type arrangement of pilots. Simulation results show that BER for OFDM can be reduced by this technique. For higher data rates we need higher value of SNR to achieve the same BER as compared to low data rate modulation schemes. As a result, the performance of OFDM system is improved in terms of BER.

The proposed technique can be used for other systems, such as single carrier system and with different modulation schemes.

REFERENCES