Power Optimization of an Iterative Multiuser Detector for Turbo CodedCDMA

Prof.Seema P. Mishra, Dr.H.N Pratihari and Prof.Swati M. Patil

ABSTRACT- **We utilize Extrinsic Information Transfer (EXIT) charts to optimize the power allocation in a multiuser CDMA system. We investigate two methods to obtain the optimal power levels: the first minimizes the total power; the second minimizes the area between the transfer curves of the interference canceller (IC) or turbo decoder. We show through simulation that the optimized power levels allow for successful decoding of heavily loaded systems. The optimal decoding schedule is derived dynamically using the power optimized EXIT chart and a Viterbi search algorithm. Dynamic scheduling is shown to be a more flexible approach which results in a more stable QoS for a typical system configuration than one-shot scheduling, and large complexity savings over a receiver without scheduling. We propose dynamic decoding schedule optimization to fix the problem, that is, on each iteration of the receiver derive the optimal schedule to achieve a target bit error rate using a minimum number of turbo decoder iterations.**

Keywords **– Channel Coding and Decoding, Turbo Code, IMUD Receiver**

I. INTRODUCTION

The advantage of the turbo decoding algorithm for parallel concatenated codes, a decade ago ranks among the most significant breakthroughs in modern communications in the past half century: a coding and decoding procedure of reasonable computational complexity was finally at hand offering performance approaching the previously elusive Shannon limit, which predicts reliable communications for all channel capacity rates slightly in excess of the source entropy rate. The practical success of the iterative turbo decoding algorithm has inspired its adaptation to other code classes, notably serially concatenated codes, and has rekindled interest in low-density parity-check codes, which give the definitive historical precedent in iterative decoding. The serial concatenated configuration holds particular interest for communication systems, since the “inner encoder” of such a configuration can be given more general interpretations, such as a “parasitic” encoder induced by a convolutional channel or by the spreading codes used in CDMA. The corresponding iterative decoding algorithm can then be extended into new arenas, giving rise to turbo equalization or turbo CDMA, among doubtless other possibilities. Such applications demonstrate the power of iterative techniques which aim to jointly optimize receiver components, compared to the traditional approach of adapting such components independently of one another.

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Algorithms are often developed and tested in floating-point environments on GPPs in order to show the achievable optimal performance. Besides shortest development time, there are no requirements on, for example, processing speed or power consumption, and hence this platform is the best choice for the job. However, speed or power constraints might require an implementation in more or less specialized hardware. This transition usually causes many degradations, for example, reduced dynamic range caused by fixed-point arithmetic, which on the other hand provides tremendous reduction in implementation complexity.

II.CHANNEL CODING AND DECODING

This chapter deals with basics of channel coding and its decoding algorithms. Following is a brief description of the simple communication model that is assumed in the sequel. This model also helps to understand the purpose of channel coding. Then, two popular coding approaches are discussed more thoroughly: convolutional coding together with Gray-mapped signal constellations and set-partition coding. Decoding algorithms are presented from their theoretical background along with a basic complexity comparison.
Consider the block diagram of the simplified communication system in Figure 2.1. It consists of an information source (not explicitly drawn) that emits data symbols $\{u_k\}$. A channel encoder adds some form of redundancy, possibly jointly optimized with the modulator, to these symbols to yield the code symbol sequence $\{c_k\}$, where ck denotes a Mary transmission symbol. Linear modulation is assumed, that is, modulation is based on a linear superposition of (orthogonal) pulses. The signal sent over the channel is therefore

$$s(t) = \sum_k c_k \cdot w(t - kT_s),$$

Where*w*( ) is the pulse waveform and $T_s$is the symbol time. The waveform channel adds uncorrelated noise *n(t)* to the signal, which results in the waveform *r(t)* at the receiver. For the remainder, the disturbance introduced by the channel is assumed to be additive white Gaussian noise (AWGN). That is,

$$\mathcal{E}\{n(t)\} = 0$$
$$\mathcal{E}\{|n(t)|^2\} = N_0/2.$$

The received waveform *r*(*t*) is demodulated to yield a discrete sequence of (soft) values *{yk}*. Based on these values, the channel decoder puts out an estimate *{ˆuk}* for the data symbols *{uk}*.

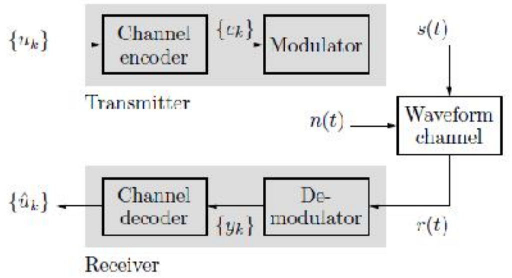

Figure 2.1: A simplified communication system.

According to Shannon, reliable communication with arbitrarily low bit error rate (BER) in the AWGN channel can be achieved for transmission rates below

$$C = \frac{1}{2}\log_2\left(1 + \frac{2E_s}{N_0}\right) \quad \text{(bits/dimension)}.$$

If there are J orthogonal signal dimensions per channel use, the transmission rate of a (coded) communication system is defined as

$$R_d = \frac{\log_2 \mathcal{M}}{J} \cdot R_c \quad \text{(bits/dimension)}, \qquad (2.1)$$

where*M* is the number of possible symbols per channel use and R_c<1 denotes the code rate of the channel code in data bits/code bits.

III.TURBO CODES

In information theory, turbo codes (originally in French Turbo codes) are a class of high-performance forward error correction (FEC) codes developed in 1993, which were the first practical codes to closely approach the channel capacity, a theoretical maximum for the code rate at which reliable communication is still possible given a specific noise level. Turbo codes are finding use in (deep space) satellitecommunications and other applications where designers seek to achieve reliable information transfer over bandwidth- or latency-constrained communication links in the presence of data-corrupting noise. Turbo codes are nowadays competing with LDPC codes, which provide similar performance.

A Soft decision approach-

The decoder front-end produces an integer for each bit in the data stream. This integer is a measure of how likely it is that the bit is a 0 or 1 and is also called *soft bit.* The integer could be drawn from the range [-127, 127], where:

- -127 means "certainly 0"
- -100 means "very likely 0"
- 0 means "it could be either 0 or 1"
- 100 means "very likely 1"
- 127 means "certainly 1"
- etc.

This introduces a probabilistic aspect to the data-stream from the front end, but it conveys more information about each bit than just 0 or 1.

IV. SYSTEM DESCRIPTION

A Existing system:

The turbo decoding algorithm for error-correction codes is known not to converge, in general, to a maximum likelihood solution, although in practice it is usually observed to give comparable performance. The quest to understand the convergence behavior has spawned numerous inroads, including extrinsic information transfer (or EXIT) charts, density evolution of intermediate quantities, phase trajectory techniques, Gaussian approximations which simplify the analysis, and cross-entropy minimization, to name a few. Some of these analysis techniques have beenapplied with success to other configurations, such as turbo equalization. Connections to the belief propagation algorithm have also been identified, which approach in turn is closely linked to earlier work [6] on graph theoretic methods. In this context, the turbo decoding algorithm gives rise to a directed graph having cycles; the belief propagation algorithm is known to converge provided no cycles appear in the directed graph, although less can be said in general once cycles appear. Interest in turbo decoding and related topics now extends beyond the communications community, and has been met with useful insights from other fields; some references in this direction include which draws on nonlinear system analysis, which draws on computer science, in addition to (predating turbo codes) and (more recent) which inject ideas from statistical physics, which in turn can be rephrased in terms of information geometry. Despite this impressive pedigree of analysis techniques, the “turbo principle” remains difficult to master analytically and, given its fair share of specialized terminology if not a certain degree of mystique, is often perceived as difficult to grasp to the non-specialist. In this spirit, the aim of this paper is to provide a reasonably self-contained and tutorial development of iterative decoding for parallel and serial concatenated codes. The paper does not aim at a comprehensive survey of available analysis techniques and implementation tricks surrounding iterative decoding, but rather chooses a particular ad- vantage point which steers clear of unnecessary sophistication and avoids approximations.

B Proposed system:

The project work focuses on joint optimization of the power and decoding schedule is prohibitivelycomplex so we break the optimization in two parts and first optimize power levels of each user then optimize the decoding schedule using the optimized power levels. Large gainsin power efficiency and complexity can be achieved simultaneously. Furthermore, our optimized receiver has a lower convergence threshold and requires less iterations to achieve convergence than a conventional receiver. We show that our proposed optimization results in a more consistent quality of service (QoS).

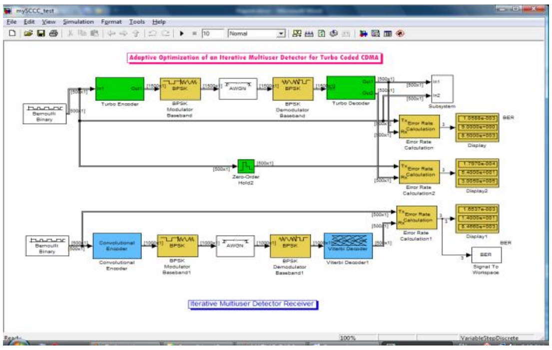

Fig.4.1. IMUD receiver with control blocks

The major advantage of dynamic scheduling over static scheduling is that the method compensates for performance better/worse than expected (average) due to differences in channel conditions Over decoding blocks, or differences in the decoding trajectory. Using dynamic scheduling we have a more reliable receiver or similar complexity.

V.IMPLEMENTATION

Implementation of any software is always preceded by important decisions regarding selection of the platform, the language used, etc. these decisions are often influenced by several factors such as real environment in which the system works, the speed that is required, the security concerns, and other implementation specific details. There are three major implementation decisions that have been made before the implementation of this project. They are as follows:

1. Selection of the platform (Operating System).
2. Selection of the programming language for development of the application.
3. Coding guideline to be followed.

RESULTS:

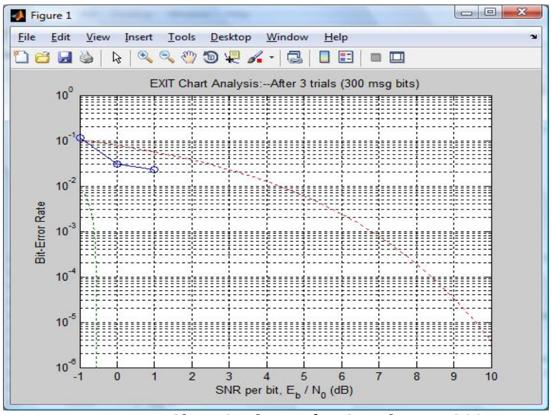

Fig 5.1 EXIT Chart Analysis after 3 trial using 300 message bits

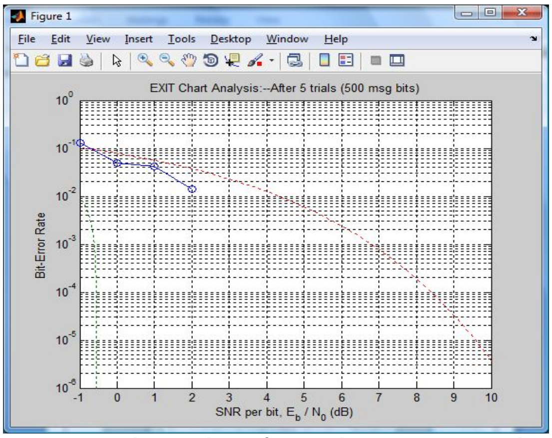

Fig5. 2 EXIT Chart Analysis after 5 trial using 500 message bits

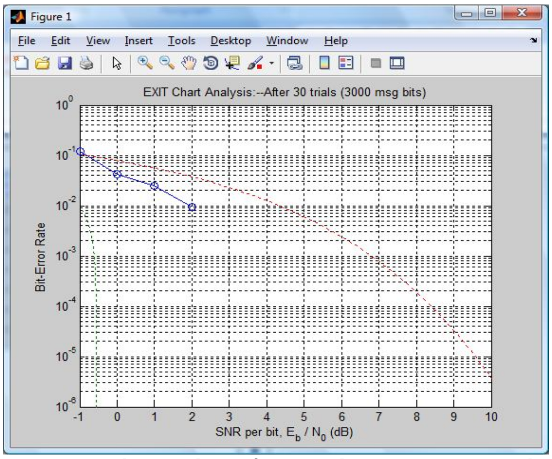

Fig 5.3 EXIT Chart Analysis after 30 trial using 3000 message bits

VI.CONCLUSION

We have optimized a turbo MUD receiver for unequal power turbo-coded CDMA system through EXIT chart analysis. The results in prior works were used to derive *effective* EXIT functions for FEC decoders and an interference canceller which enabled analysis of the system as in the equal power case. We utilized a nonlinear constrained optimization as in prior work to optimize the power levels of groups of users in the system. We modified the algorithm proposed in prior work to dynamically derive the optimal decoding schedule for the IMUD receiver. We then showed through simulation that this power optimized system using dynamic scheduling achieves similar BER performance as a conventional receiver with significant complexity savings. Furthermore it outperforms the statically derived optimal schedule through reducing the variance of the per packet BER. We also proposed a method for estimating the SNR in an AWGN CDMA channel and showed that power and schedule may be optimized without any trade-off. Finally, we determined that a combination of static and dynamic scheduling offers the best benefit for the cost.

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Study of the Variation of Power Loss with Frequency along a Rectangular Waveguide for TE $_{10}$ mode due to Conductor Attenuation

Biswajit Ghosh

Abstract: **Nowadays microwave communication is an important part in modern telecommunication systems. Microwave travels through waveguides. So loss within a waveguide is a vital matter of concern. In this paper, successfully observed a rectangular waveguide and its powerloss i.e. attenuation variation with the change of microwave frequencies(X-Band) for TE_{10} mode. A simulation is also done using MATLAB platform. For this the b/a ratio of the rectangular waveguide is selected to be 0.5 and also found out the result with other values b/a ratios of 0.2, 0.8.**

Keywords: Rectangular Waveguide, TE_{10} mode, Power Loss variation, X-band Frequency

I. Introduction

Losses in two conductor lines increase with frequency.Hence Microwaves are transmitted through hollow metallic tubes (waveguides) of uniform cross section with minimum loss[1]. If the waveguide walls are not perfectly conducting, there exists a tangential component of electric field and normal component magnetic field at the walls. This results in average power delivered to the waveguide walls and dissipated in it and hence continuous attenuation of propagating electromagnetic waves.[2] Here we will study the variation of conductor attenuation with frequency.

II. Conductor Attenuation In Rectangular Waveguides

If $\vec{H}_t$ and $\vec{E}_t$ are the tangential components of electric and magnetic fields then power loss per unit guide length is given by

$$P_L = \frac{1}{\Delta z}\frac{1}{2}\mathrm{Re}\oiint_{S1}\left(\vec{E}_t \times \vec{H}_t^*\right)d\vec{S}_1 \qquad (1)$$

Δz is the length element

(1)

Since $\vec{E}_t$ results from the flow of current due to $\vec{H}_t$ and the direction of current flow is normal to $\vec{H}_t$,hence $\vec{E}_t$ and $\vec{H}_t$ must be at right angles[8]. The ratio of tangential magnetic field may be defined as high frequency conductor impedance The power loss per unit length of the guide is the space rate of decrease of power flow, or

$$P_L = \alpha_c Z_o \oiint_{S1}|H_o|^2 e^{-2\alpha_c z} ds \qquad (2)$$

Above equation is the fundamental equation and may be used to evaluate attenuation arising from imperfect conducting walls. However, the problem of evaluating P_L seems to be a difficult one at the outset because the loss depends upon the field configuration within the guide , and the field configuration in turn, depends to some extent ,upon the losses[3][7]. We first write the equations for the electric and magnetic fields as though the guide were loss-less. Since the walls of the guide are assumed to be perfectly conducting, the tangential component of the electric field at the guide walls in zero[5]. However, there will be a tangential component of magnetic intensity which is terminated by a current flowing on the surface of the conductor. If we now assume that the guide walls are imperfectly conducting, the tangential component of magnetic and hence P_L may be evaluated[6].

We find that the attenuation constant arising from guide walls is given by

$$\alpha_c = \frac{R_S \oiint_{S1}|H_t|^2 ds_1}{2Z_o \oiint_{S1}|H_T|^2 ds} \qquad (3)$$

The numerator of above equation is evaluated at each conductor surface whereas in the denominator it is over cross-section of the waveguide[4].

We now proceed to find α_c for few waveguide modes.For TM_{mn} Mode since $H_{z=0}$ in this case,

$$\alpha_{c(TM)} = \frac{R_S \lambda_c^{\,2}\left(\dfrac{n^2 a}{b^2} + \dfrac{m^2 b}{a^2}\right)}{2\eta ab\sqrt{1-(\lambda/\lambda_c)^2}} \qquad (4)$$

Or, if we use

$$\lambda_c = \frac{2ab}{\sqrt{n^2a^2 + m^2b^2}}$$

$$\alpha_{c(TM)} = \frac{2R_s}{b\eta\sqrt{1-(f_c/f)^2}}\,\frac{m^2(b/a)^3 - n^2}{m^2(b/a)^2 + n^2} \qquad (5)$$

Similarly, the expression for $\alpha_{c(TM)}$ may be obtained as

$$\alpha_{c(TE_{mn})} = \frac{2R_s}{b\eta\sqrt{1-(f_c/f)^2}}\left[\left(1+\frac{b}{a}\right)\left(\frac{f_c}{f}\right)^2 + \left\{1-\left(\frac{f_c}{f}\right)^2\right\}\left\{\frac{\frac{b}{a}\left(\frac{b}{a}m^2+n^2\right)}{\left(b^2m^2/a^2\right)+n^2}\right\}\right] \qquad (6)$$

For TE_{10} mode [2],[4] carriers and b/a=2 (substitute m=0 and n=1 in Eq.6)we get,

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$$\alpha_{c(TE_{10})}=\frac{R_S\left[1+\frac{2b}{a}\left(\frac{f_c}{f}\right)^2\right]}{\eta b\sqrt{1-\left(f_c/f\right)^2}} \quad (7)$$

III. Simulated Results Of Waveguide Attenuation Variation With Frequency For A Particular Mode (TE_{10} Mode)

Based on the theoretical discussion I have done a MATLAB program to find out the attenuation of a particular mode(TE_{10})as it flows along the waveguide. For this I have selected a b/a ratio of the rectangular waveguide equals to 0.5 and also found out with other values b/a ratios of 0.2, 0.8. I have varied the frequency from 6.5 GHz to 14 GHz [f_c =6.562GHz].

The simulated graph is shown below:

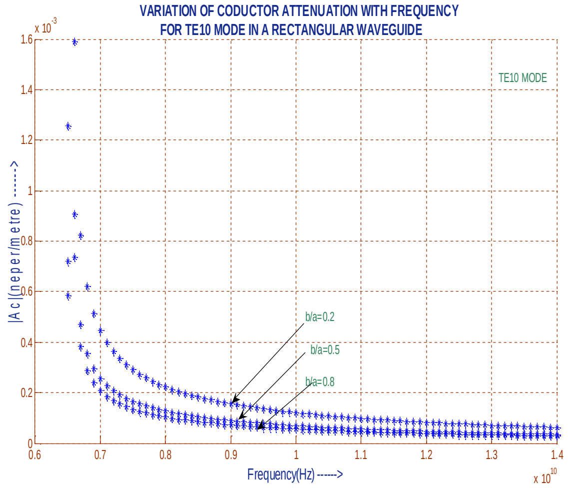

Fig.1 Variation of conductor attenuation with frequency for TE_{10} mode in a rectangular waveguide for different b/a ratios

IV. Experimental Analysis Of Waveguide Power Losses Variation With Frequency For A Particular Mode(TE_{10} Mode)

A experimental demonstration was then performed to analyse the conductor attenuation that occurs as an transverse electromagnetic wave propagates within the waveguide. So a experiment in the microwave laboratory was done, the details of which is given below-

A. *Apparatus used:*

Gunn oscillator, Isolator, Pin modulator, Frequency Meter Variable Attenuator, Rectangular waveguide (Brass Material)Matched Termination, Tunable Detector, Power Meter, CRO,SWR Meter, Coaxial to waveguide adapter

B. *Procedure*

The main aim was simply to find variation of output power with frequency. For that purpose I chose the Gunn oscillator as a source of microwave power. This microwave power is then fed to the waveguide section as shown in the fig.2 below:

The Gunn bias was set at 8.7mV. In the slotted line section, the probe was tuned for maximum reading in dB on SWR meter. Then I tuned the frequency meter to get a 'dip'(minimum reading) on CRO display and note down the frequency directly from frequency meter. Next I detuned the DRF Meter .We confirm this frequency also by using the slotted line method of measuring frequency from the relation $\lambda_g=2(d_1-d_2)$and thereby from eqn

$$f=\frac{c}{\lambda_0}=c\sqrt{\frac{1}{\lambda_c^{\,2}}+\frac{1}{\lambda_g^{\,2}}} \quad (8)$$

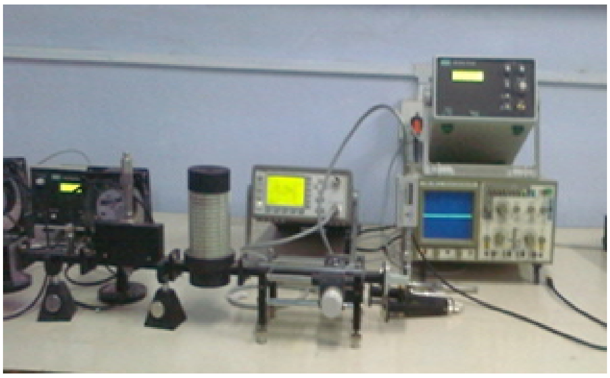

Fig.2 Power Loss measurement setup

A waveguide to coaxial adapter was then substituted in place of matched termination and connected it to the power meter. The power was directly observed from the power meter at that frequency. The steps above are repeated and we observe the power variation for X-band frequency of operation and b/a ratio=0.5. The change of power output to the power input gives the power loss. I plotted the Power loss Vs Frequency graph. The freq and powerloss values I measured is given below-

TABLE I
POWER LOSS VS FREQUENCY VARIATION

Frequency (GHz)	Power Loss (dBm)
11.4	29.67
11.2	26.9
11.02	25.98
10.86	29.07
10.68	30.9
10.59	30.64
10.41	35.4
10.32	35
10.23	35.1
10.08	35.12
9.97	35.21
9.97	35.21
9.7	35.55
9.26	38
9.049	53.53

9.038	57.83
9.026	62.16
9.02	61.9
9.007	61.15
8.94	61.37
8.91	61.37

After placing this readings , the graph obtained is given below:

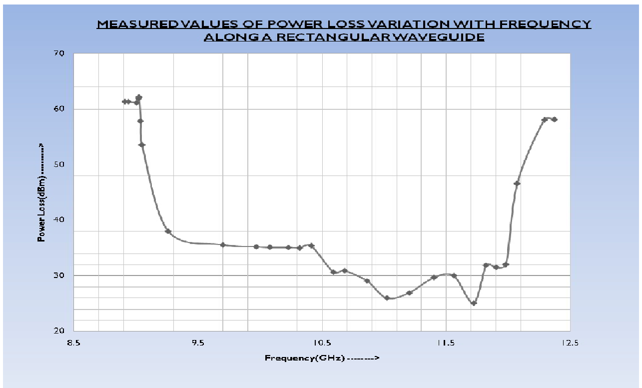

Fig.3 Measured values of Power Loss variation with frequency along a rectangular waveguide

V. CONCLUSIONS

This graph does not have much of deviation from the graph obtained from the MATLAB programming except a little change due to the experimental error of human and also due to those instruments used. Now as we can see from the graph that the power loss within the wave guide is minimum when the frequency ranges from 9.5 to 12 GHz. More precisely we can say at 11.7 GHz frequency it meets minimum losses .The losses encountered accounts for the various attenuation losses that occurs within the waveguide. Here since the dielectric is air ($€_r$=1),we can say this attenuation is primarily the conductor attenuation.

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Outperformance of Proactive Spectrum Handoff in Cognitive Radio ADHOC Network

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ABSTRACT- Cognitive Radio (CR) is the modest technology to make the unlicensed users to make use of licensed spectrum in the opportunistic manner. Collision and throughput are the main parameters to be checked while performing handoff. Proactive Spectrum is proposed to address these Concerns. Various Channel sensing methodologies were simulated and comparison was made between proactive, Reactive Spectrum. Simulation results show that our proactive spectrum handoff outperforms the reactive spectrum handoff and wireless networks approach in terms of higher throughput and fewer collisions to licensed users.

KEY Words- Congestion, CR, Handoff and proactive spectrum,CSMA/CD

1. INTRODUCTION

An adhoc network is a collection of wireless mobile hosts forming a temporary network without the aid of any established infrastructure or centralized administration. Each node is considered to be alike here. It is needed to introduce some intelligence to the adhoc networks in order to improve their throughput efficiency. The concept of Cognitive Radio (CR) has been employed to achieve this. CR enabled devices are 'clever' and can listen to the surrounding wireless environment and can select appropriate frequency band, modulation scheme with various parameters and signal space or specific power level as per the requirement. COGNITIVE Radio is a technology evolved from software defined radio (SDR) for wireless communications in which a network changes its transmission or reception parameters to communicate efficiently avoiding interference with licensed or unlicensed users [1].It is generally used for military and civilian applications. Cognitive radio ad hoc networks (CRAHNs) constitute a viable solution to solve the current problems of inefficiency in the spectrum allocation, and to deploy highly reconfigurable and self-organizing wireless networks. It is needed to introduce some intelligence to the adhoc networks in order to improve their throughput efficiency.

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Cognitive radio (CR) devices are envisaged to utilize the spectrum in an opportunistic way by dynamically accessing different licensed portions of the spectrum [1]. The limited available spectrum and the inadequacy in the spectrum usage necessitate a new communication standard to utilize the existing wireless spectrum opportunistically [2]. Here, we discuss the mechanisms that can be followed to provide better efficiency in utilizing an existing cognitive network. This methodology also ensures that the existing channel frequencies are not wasted and the frequencies that are in use are utilized to the maximum extent without the need for bothering about the effects of interference [3].

2. COGNITIVE RADIO

Cognitive radio is an emerging and promising technology for getting the most out of consumption of the limited radio bandwidth while accommodating the increasing amount of network services and applications in wireless network techniques. The cognitive radio networks are almost done for dynamic spectrum allocation to access the networks. [1] The key features of cognitive radio network are wideband signal processing techniques for digital radio, advanced wireless communications methods, artificial intelligence and machine learning techniques, and cognitive radio-aware adaptive wireless/mobile networking protocols.

A cognitive radio is a kind of two-way radio that automatically changes its transmission or reception parameters, in a way where the entire wireless communication network of which it is a node communicates efficiently, while avoiding interference with licensed or licensed exempt users. This alteration of parameters is based on the active monitoring of several factors in the external and internal radio environment, such as radio frequency spectrum, user behavior and network state. Fig 2.1 (17) shows the sample CR network.

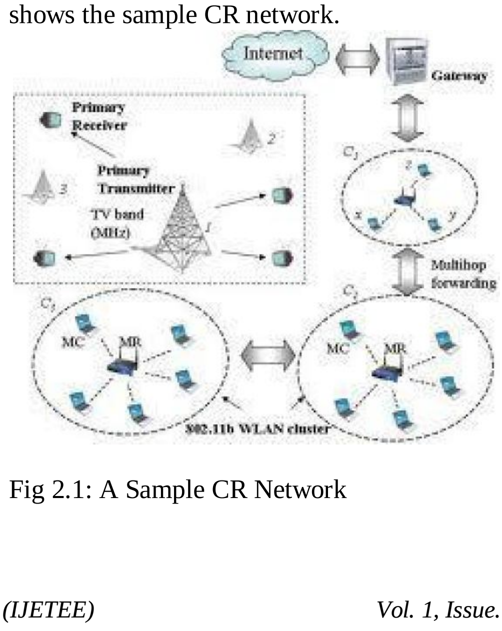


Fig 2.1: A Sample CR Network

Regulatory bodies in various countries (including the Federal Communications Commission in the United States and Ofcom in the United Kingdom) found that most of the radio frequency spectrum was inefficiently utilized. For example, Cellular network bands are overloaded in most parts of the world, but many other frequency bands, such as military, amateur radio and paging frequencies are not. Independent studies performed in some countries confirmed that observation, and concluded that spectrum utilization depends strongly on time and place [4]. Moreover, fixed spectrum allocation prevents rarely used frequencies (those assigned to specific services) from being used by unlicensed users, even when their transmissions would not interfere at all with the assigned service. This was the reason for allowing unlicensed users to utilize licensed bands whenever it would not cause any interference (by avoiding them whenever legitimate user presence is sensed). This paradigm for wireless communication is known as cognitive radio. Since the spectrum present is limited, all users cannot be allowed to access this network (18).

2.1 Categories of CR

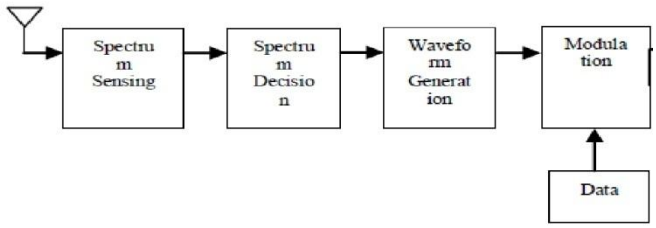
The users are divided into two basic categories.

Primary Users: Since the spectrum present is limited, all users cannot be allowed to access this network of any other unlicensed users. Primary users do not need any modification or additional functions for coexistence (14).

Secondary Users or Unlicensed Users: They access the licensed spectrum as a visitor, by opportunistically transmitting on the spectrum holes.

2.2 Waveforms of Cognitive Radio:

The most important job of the Cognitive Radio (CR) is to efficiently use spectrum hole which is assigned to a primary user (PU) or licensed user. In order to achieve this goal, CRs have to detect the reappearance of PU frequently. They should quit the spectrum immediately as soon as a PU is detected in order to minimize their reciprocal interference. This suggests that CR has to change it's transmitting waveform and adapt to the spectrum environment. Therefore, the adaptive waveform [10] techniques have been investigated. The term adaptive waveform stands for ―A time domain pulse in the radio frequency (RF) range that has the desired frequency response [09]. In this technique, CRs will periodically monitor the RF spectrum (spectrum sensing) and choose the best available spectrum allocation (spectrum decision). On basis of the spectrum information obtained, CRs generate an adaptive carrier waveform which fits only the free band. As soon as the waveform is generated, digital data will be modulated using this waveform and transmitted. Fig 2.2 in (1) shows the process of the adaptive waveform generation.



Obviously, how to decide and select a waveform for transmitting based on environmental measures is one of the most important problems for CR. A new on adaptive carrier waveform scheme is proposed in [10] to adapt to any band without bringing about harmful interference. It is useful in accessing TV spectrum with high spectrum utilization efficiency.

3. SPECTRUM SENSING IN COGNITIVE RADIO SYSTEMS

Spectrum sensing is basically a binary hypothesis testing problem and the key performance measures for a spectrum sensing method are the probability of correct detection, probability of false alarm, and probability of miss. Cooperative spectrum sensing in CR would be challenged by some uncertainties such as channel fading or shadowing. The low received signal strength is not enough for CR to detect whether the primary user exists or not. The impact of noise Spectrum uncertainty is also important. A prior knowledge of noise is required during sensing, but it's not available in practice. Because of such channel uncertainties, spectrum sensing must be sensitive enough to overcome such uncertainties. On the other hand, hidden terminal problem is also a threat to CR. If a secondary node is out of the range of primary transmitter and may generate false decisions, it will interfere the primary user. Cooperative spectrum sensing [2] is a possible way to solve the threats. By cooperation within secondary users, more information could be taken during spectrum sensing. This kind of sensing technique can be implemented either in a centralized or in a distributed way. [3][4]

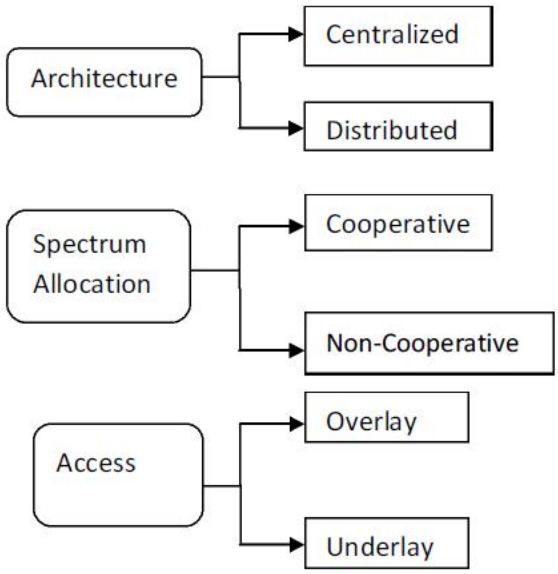


Fig 3.1: Architecture of CR Network

3.1 Centralized Cooperative Spectrum Sensing:

In the centralized approach, as illustrated in Fig 3.1, the CR base station receives all the sensing information from the secondary users and determines the state of spectrum. Then the local results are sent to a CR base station that performs data fusion and determines the final spectrum sensing result. The sensing terminals may return different results due to some reasons such as the distance between the primary user and the sensing terminals, shadowing, or fading. Each time a secondary user requests an access of a certain spectrum, a permission of the CR base station is needed. The cost of this scheme might be expensive, but it's effective to manage to spectrums and increase the correction of sensing.

3.2 Distributed Cooperative Spectrum Sensing

On the other hand, distributed approach requires exchange of observations among all the secondary users (12). Each secondary user plays a role of fusion center, and chooses the best channel from the available spectrum, based on the local data available to them (often from the neighbor nodes). This scheme could be easier implemented with low cost, but it is shared to manage and some collision might be happened.

Random channel selection: A SU randomly Chooses a channel from its predicted available channels. Greedy channel selection: In this method, only one pair of SUs is considered in the network. The SUs can obtain all the channel usage information and predict the service time on each channel. Thus, when a spectrum handoff occurs, a SU selects a pre-determined channel that leads to the minimum service time Local bargaining: In this method, SUs form a local group to achieve collision free channel assignment. To make an agreement among SUs, a four-way handshake are needed between the neighbors (i.e., request, acknowledgment, action, acknowledgment). Since one of the SUs is the initiating node which serves as a group header, the total number of control messages exchanged is 2NLB, where NLB is the number of SUs need to perform spectrum handoffs. Since for channel selection schemes, reducing the number of collisions among SUs is the primary goal, we consider the SU throughput, average SU service time, collisions among SUs, and average spectrum handoff delay as the performance metrics.

3.3 Advantages:

1. A distributed channel selection scheme (12) to eliminate collisions among unlicensed users in a multiuser spectrum handoff so there is no interference or collisions.
2. Due to no collisions the proactive spectrum can achieve high throughput value and higher packet delivery.
3. Due the spectrum handoff packet loss is greatly reduced.
4. Compare to reactive spectrum the quality of service is improved.

4. PROACTIVE SPECTRUM HANDOFF PROTOCOL

In this section, we first propose the spectrum handoff criteria and policies that a CR transmitting pair is required to follow. Then, the details of the proposed spectrum handoff protocol are presented.

4.1 Proposed Spectrum Handoff Criteria and Policies:

By utilizing the sensed channel usage statistics, an SU can make predictions of the channel availability before the current transmission frame ends. Based on the prediction, the SU decides whether to stay in the present channel, or switch to a new channel, or stop the on-going transmission. We propose two criteria for determining whether a spectrum handoff should occur: 1) The predicted probability that the current and a candidate channel are busy or idle and 2) the expected length of the channel idle period. Based on these criteria, we design spectrum handoff policies.

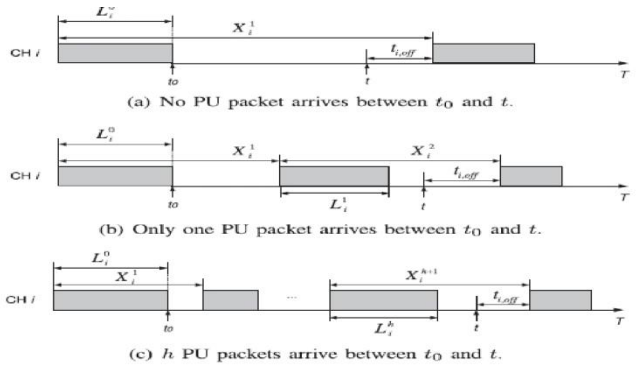


Fig 4.1: The PU activity of channel 1

Fig 4.1 in (12) shows the PU traffic activity on channel i, where X_{ki} represents the inter arrival time of the k_{th} packet. We denote Y(t) as the number of PU packets that arrive between t_0 and t. As shown in Fig 4.1a, the probability that channel i is idle and no PU packet arrives between t_0 and t is given by

$$Pr\ (N_i\ (t) = 0,\ Y\ (t) = 0) = Pr\ (X^1_i > t - t_0 + L^0_i)$$

Where L_{ki} denotes the length of the k^{th} PU data packet on channel i. As shown in Fig 4.1b, the probability that channel i is idle and only one PU packet arrives between t_0 and t is given by

$$Pr\ (N_i\ (t) = 0,\ Y\ (t) = 1)$$

$$= Pr\ (X^l_i + L^l_i < t - to + L^O_i)\ Pr\ (X^l_i + X^2_i > t - to + L^O_i)$$

Similarly, in Fig 4.1c, the probability that channel i is idle and h (h 2 ½1; U) PU packets arrive, where U is the maximum number of PU packets that could arrive between t_0 and t, is

$$Pr\ (\ N_i\ (t) = 0, Y(t) = h)$$

$$= Pr(\sum_{k=1}^{h} X^k + L^h < t - t_0 + L^0)$$

$$Pr(\sum_{k=1}^{h+1} X^k < t - t_0 + L^0$$

Thus, if the PU traffic model is known and the channel statistics (e.g., PU packet arrival rate, PU packet length) are obtained from the scanning radio, the predicted probabilities can be calculated. Hence, based on the above prediction, the policy that an SU should switch to a new channel is

$$\Pr(N_i(t)) \geq T_L$$

Where T_L is the probability threshold below which a channel is considered to be busy and the SU needs to carry out a spectrum handoff, that is, the current channel is no longer considered to be idle at the end of the frame transmission. In addition, the policies that a channel j becomes a candidate channel at time t are

$$\{\Pr((N_j(t) = 0 \geq T_H \Pr(t_{j,off} > N_j(t) = 0)))\} \geq \alpha\Theta$$

where T_H is the probability threshold for a channel to be considered idle at the end of the current frame, Θ is the probability threshold α for a channel to be considered idle for the next frame transmission. The second criterion in (14) means that, in order to support at least one SU frame, the probability that the duration of the idleness of channel j to be longer than a frame size.

4.2 Proposed Spectrum Handoff Protocol:

In proposed Spectrum handoff the secondary user is made to vacate the channel before the primary user arrives .Spectrum handoff frame can be designed by using multiple channel multi access protocol whereas the single access scheme is used in Bluetooth and multiple access scheme using pseudorandom sequence made our protocol highly reliable and highly securable. The goal of our proposed protocol is to determine whether the SU transmitting pair needs to carry out a spectrum handoff and then switch to a new channel by the time a frame transmission ends by introducing the hopping pattern.

The proposed spectrum handoff protocol is based on the above proposed spectrum handoff policies. It consists of two parts. The first part, namely Protocol 1 describes how a Secondary user pair initiates a new transmission. Regardless of the coordination schemes used during channel hopping, if a data packet arrives at an SU, the SU predicts the availability of the next hopping channel (in the single rendezvous coordination scheme case) or the hopping channel of the receiver (in the multiple rendezvous coordination scheme case) at the beginning of the next slot. Based on the prediction results, if the channel satisfies the policies in (14) for data transmissions, the channel is considered available. Then, if the CTS packet is successfully received by the SU transmitter, the two SUs pause the channel hopping and start the data transmission on the same channel in the next time slot. Note that if more than one pair of SUs contends the same hopping channel for new data transmissions, only the SU pair who exchanges the RTS/CTS packets first claims the channel. Hence, no RTS collision will occur. The second part, namely, Protocol 2 is on the proactive spectrum handoff during an SU transmission. Fig4.2 in (4) illustrates the operations of Protocol 2.

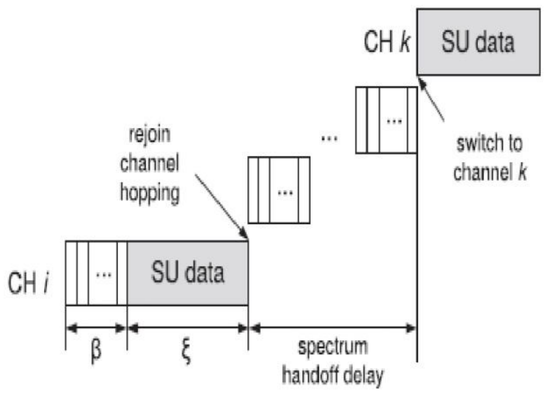


Fig 4.2: Operation of protocol

Using the proposed protocol, the SU transmitting pair can avoid disruptions with PUs when PUs appears. Based on the sensed channel usage information, an SU transmitter checks the spectrum handoff policy in (6) for the current channel by predicting the channel availability at the end of the frame.

5. CONGESTION DETECTION

The dual buffer thresholds (13) and weighted buffer difference are used to detect the congestion. The Fig.5.1 shows the details of buffer state such as “accept state (0 - Q_{min})”, “filter state (Q_{min} - Q_{max})” and “reject state (Q_{max}- Q)”. The different buffer states are reflected different channel loading which is used to accept or reject packets in different states.

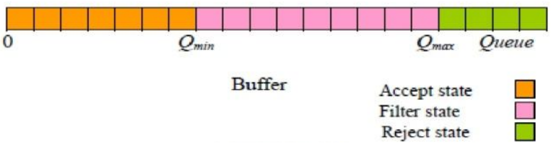


Fig. 5.1 Buffer state

The packet at each node has to send for buffer monitoring and piggybacks its weighted buffer changing rate (*WR)* and weighted queue length (*WQ)* with outgoing packets. The corresponding congestion level bit in the outgoing packet header is set if a node’s buffer occupancy exceeds a certain threshold and its packets has higher priority among neighborhood. The weighted buffer with length WQ (t) after Δt and the weighted buffer difference at time t + Δt are calculated as,

$$WQ(t + \Delta t) = WQ(t) + WR * \quad t \quad (1)$$

$$WQD_{nodei}(t + \Delta t) = \sum_{j=1}^{N} \mathrm{DP}(\mathrm{packet}_j) - \mathrm{Max}(WQ_k(t + \Delta t)) \quad (2)$$

Where $k_{\in}$ neighbor ($node_i$) and N is the total number of packets in the buffer. If $WQD_{nodei}(t+\Delta t) \geq 0$, the data of node i is the most important among its neighbors. If congestion happens, other nodes should lower down their data sending rate to mitigate node’s congestion.

6. SIMULATION RESULTS

The current simulation is carried out in the network simulator (Ns-2). This is very helpful in the networking concepts. By using this, the parameters like throughput, delay, collision rate, packet drop can be measured.

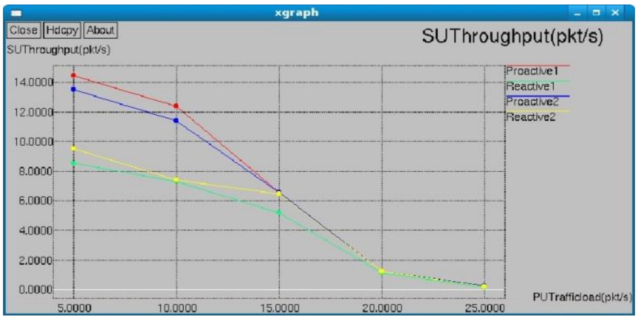


Fig 6.1 Comparison of proactive Vs Reactive

From the graph, the throughput offered in the proactive will be better than the reactive spectrum is shown in fig 6.1. It clearly shows that 30% increase in throughput in proactive when compared with the reactive spectrum. The collision rate in proactive is less than the reactive spectrum is shown in fig 6.2. From graph, its shown that 27% of the collision rate gets decreased in proactive spectrum.

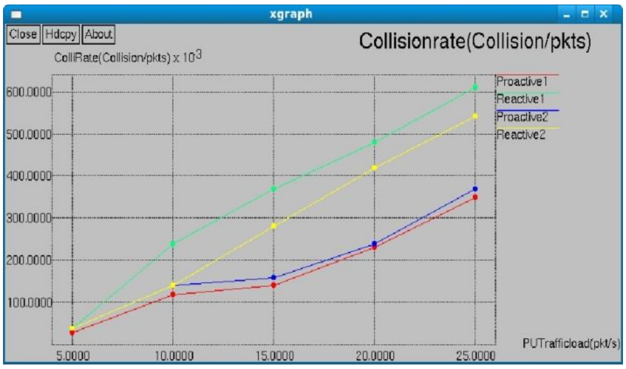


Fig 6.2: Collision Rate Vs Load

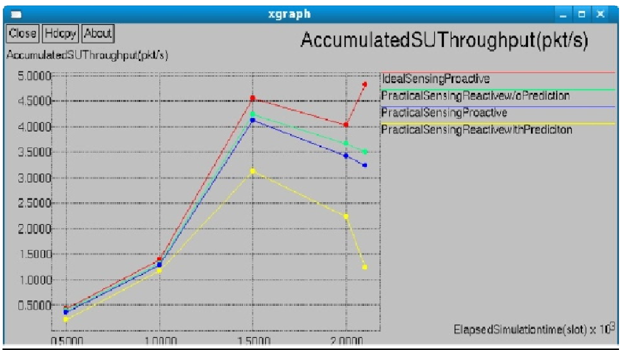


Fig6.3: Accumulated throughput Vs Elapsed time

From the graph, it is clearly shown that the ideal sensing proactive gets higher throughput than practical sensing throughput is shown in fig 6.3. The average delay on the channel is also maintaining the initial delay on greedy channel selection on comparing with the secondary user channel which is shown in fig 6.4.Among all the channel sensing methods, ideal channel sensing performs well, but it cannot be practically implemented, next to ideal sensing is that proactive sensing with prediction which outperforms better than other channel sensing techniques.

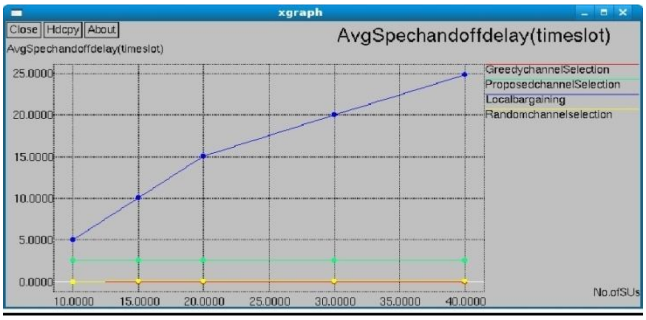


Fig 6.4: Avg delay Vs Secondary user

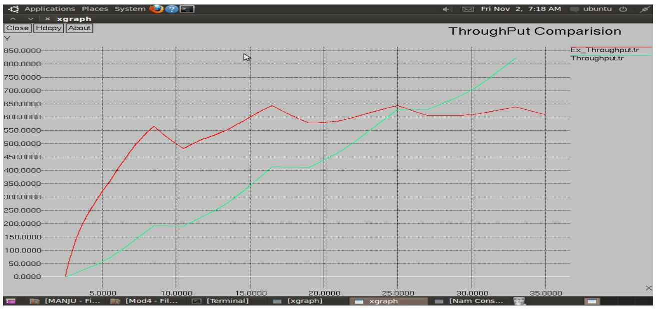


Fig 6.5: Throughput analyses in Wireless network

The throughput analyses for the same number of nodes in wireless network in Fig 6.5 shows that our proposed proactive method outperforms normal wireless network methodologies especially carrier sense multiple access/collision detection in terms of higher throughput.

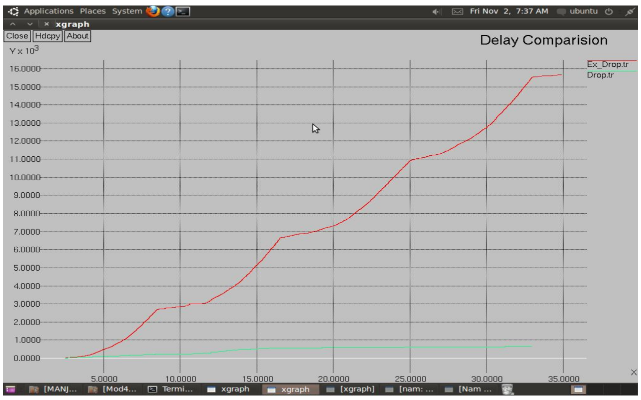


Fig 6.6: Delay analyses in Wireless network

From the Fig 6.6, it is clearly shown that the Proactive spectrum produces lesser delay, when compared to Wireless network with Collision detection technique.

Table 6.1: Comparison Chart for WN & CR

Parameter	**Wireless Network Technology**	**CR Technology**
Throughput	Low (78.76%)	High (85%)
Delay	High (5.41%)	Low (3.5%)

From the Table 6.1, it is clear that Proactive spectrum cognitive radio outperforms the wireless networks in terms of throughput and average delay to primary user by letting primary users predicting the future spectrum availability in advance and perform spectrum handoffs before it occupies the current spectrum.

7. Conclusion

In this paper, Collision avoidance is shown by using proactive spectrum handoff, proactive spectrum outperforms the reactive spectrum in terms of higher throughput and shorter Handoff delay. Compared to other channel sensing method and Collision detection in wireless networks, our method achieves fewer disruptions to primary users.

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Estimation and Optimization of Power dissipation in CMOS VLSI circuit design: A Review Paper

Hasmukh P Koringa, Prof. (Dr.) Vipul A Shah and Prof. Durgamadhab Misra

Abstract: **This paper covers critical review of various methods and techniques for estimation and optimization of power in CMOS (complementary metal oxide semiconductor) VLSI (very large scale integrated) circuit design. There is different design techniques like multiple supply, multiple threshold, multiple transistor size and stack forcing used for low power design. Linear programming algorithm used to find non-critical path node for multiple supply and multiple threshold assignment. Genetic algorithm can be used to find optimum combination of different design parameters value. In section I contain introduction and need of low power design. Different source of power dissipation discuss in section II. Section III and IV hold review concepts of theories and literature review on previous work done so far respectively. Research scope is in section V and conclusion has in VI and last references.**

Key Words: **CMOS, Estimation, Low Power Design, Optimization, VLSI**.

I INTRODUCTION

The continuing decrease in the feature size and the corresponding increases in chip density and operating frequency have made power consumption a major concern in VLSI design. Excessive power dissipation in integrated circuits discourages their use in portable systems. It also causes overheating, which degrades the performance and reduces chip lifetime. To control their temperature levels, the chips need specialized and costly packaging and cooling arrangement, which result in further escalation of the system cost. The growing need for portable communication devices and computing systems has increased the need for optimization of power consumption in the chip. Overall, low power design critical technology needed in the semiconductor industry today. Simultaneously, we also need to speed up the critical paths of the circuit, while reducing its power consumption.

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II SOURCES OF POWER CONSUMPTION IN CMOS

The power consumption in digital CMOS circuits can be calculated by:

$P_{total} = P_{dynamic} + P_{short\text{-}circuit} + P_{static} + P_{leakage}$

where P_{total} is the total power consumption, $P_{dynamic}$ is the dynamic power consumption due to switching of transistors, $P_{short\text{-}circuit}$ is the short-circuit current dissipation when there is direct path from the power source down to ground, P_{static} is the static power consumption, and $P_{leakage}$ is the power consumption due to leakage currents.

A. DYNAMIC POWER CONSUMPTION

The dynamic power consumption, $P_{dynamic,}$ is due to the charging and discharging of the parasitic capacitance in the circuit.

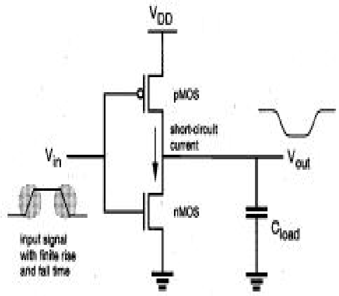

Figure.1. CMOS inverter

$$P_{dynamic} = C_L \bullet V^2_{dd} \bullet N \bullet f \qquad (1)$$

Where, C_L is parasitic and interconnect capacitance, V_{dd} is supply voltage, N is switching activity factor and f is frequency of signal [6].

B. SHORT CIRCUIT POWER

The short circuit power, $P_{short\text{-}circuit,}$ is due to direct path from the power supply to ground, during the transition phase.

$$P_{short\text{-}circuit} = k \bullet (V_{dd} - 2V_t)^3 \bullet t \bullet N \bullet f \qquad (2)$$

Where *K* is a constant that depends on the transistor size and the technology, V_t is the threshold voltage of the nMOS and pMOS transistors, t is the rise or fall time of the input signal, *N* is the average number of transitions in the inverter's output, and *f* is the clock frequency [20].

C. STATIC CIRCUIT POWER

Static power dissipation in CMOS is due to leakage currents.

D. LEAKAGE CURRENTS

There are six different leakage currents in short channel transistor illustrate in Figure.2.

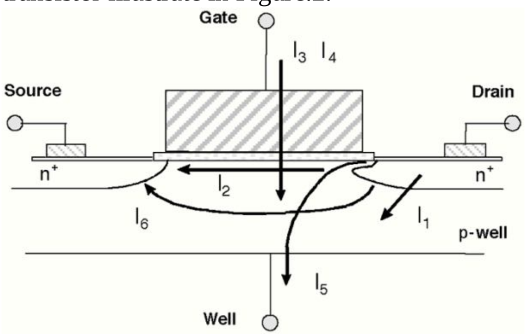

Figure. 2. Transistor leakage currents [14]

Where, I_1 is the reverse-bias pn junction leakage. A reverse-bias pn junction leakage I_1 has two main components: one is minority carrier diffusion/drift near the edge of the depletion region; the other is due to electron-hole pair generation in the depletion region of the reverse-biased junction.

$I_{leakage}=Is\bullet(e^{(Vdd/Vth)}-1)$ (3)

where I_S is the reverse saturation current, V_{dd} is the voltage bias, and $Vth = kT/q$ is the thermal voltage. For engineering purposes, we can assume that the leakage current is equal to the reverse saturation current.

The reverse saturation current is given by:

$Is=q\bullet n_i^2\bullet A\bullet(\frac{Dp}{NdWn}+\frac{Dn}{NaWp})$ (4)

where q is electron charge, n_i is the intrinsic concentration, A is area of *pn* junction diode (actually drain area), *Dp* and *Dn* are the electron and hole diffusion coefficient, *Nd* and *Na* are the donor and acceptor concentration, *Wp* and *Wn* are the width of *p*- and *n*-side of *pn* junction diode, respectively.

I_2 is the subthreshold leakage; which occurs due to carrier diffusion between the source and drain when the gate-source voltage, V_{gs}, has exceeded the weak inversion point, but is still below the threshold V_t, where carrier drift is dominant. In this regime, the MOSFET behaves similarly to a bipolar transistor, and the sub-threshold current is exponentially dependent on the gate-source voltage. The current in the subthreshold region is given by [30]:

$I_{ds}=Ke^{(Vgs-Vt)/nVth}(1-e^{(Vds/Vth)})$ (5)

where K is a function of the technology, Vth is the thermal voltage, Vt is the threshold voltage and $n = 1+Wtox/D$, where tox is the gate oxide thickness, D is the channel depletion width, $W = ESi/Eox$. For $Vds >> Vth$, $(1-e^{-Vds/Vth}) = 1$; that is, the drain to source leakage current is independent of the drain-source voltage *Vds*, for *Vds*=0.1 volt [30].

I_3 is the oxide tunneling current; Reduction of gate oxide thickness results in an increase in the field across the oxide. The high electric field coupled with low oxide thickness results in tunneling of electrons from substrate to gate and also from gate to substrate through the gate oxide, resulting in the gate oxide tunneling current.

I_4 is the gate current due to hot-carrier injection; in a short-channel transistor, due to high electric field near the Si–SiO_2 interface, electrons or holes can gain sufficient energy from the electric field to cross the interface potential barrier and enter into the oxide layer. This effect is known as hot-carrier injection. The injection from Si to SiO_2 is more likely for electrons than holes, as electrons have a lower effective mass than that of holes, and the barrier height for holes (4.5 eV) is more than that for electrons (3.1 eV).

I_5 is the GIDL (Gate Induce Drain Leakage); GIDL is due to high field effect in the drain junction of an MOS transistor. When the gate is biased to form an accumulation layer at the silicon surface, the silicon surface under the gate has almost same potential as the p-type substrate. Due to presence of accumulated holes at the surface, the surface behaves like a p region more heavily doped than the substrate. This causes the depletion layer at the surface to be much narrower than elsewhere. The narrowing of the depletion layer at or near the surface causes field crowding or an increase in the local electric field, thereby enhancing the high field effects near that region [39]. When the negative gate bias is large (i.e., gate at zero or negative and drain at V_{DD}), the n+ drain region under the gate can be depleted and even inverted. This causes more fields crowding and peak field increase, resulting in a dramatic increase of high field effects such as avalanche multiplication and BTBT [39]. The possibility of tunneling via near-surface traps also increases. As a result of all these effects, minority carriers are emitted in the drain region underneath the gate. Since the substrate is at a lower potential for minority carriers, the minority carriers that have been accumulated or formed at the drain depletion region underneath the gate are swept laterally to the substrate, completing a path for the GIDL [11]. Thinner oxide thickness and higher (higher potential between gate and drain) enhance the electric field and therefore increase GIDL.

I_6 is the channel punch through current. In short-channel devices, due to the proximity of the drain and the source, the depletion regions at the drain-substrate and source-substrate junctions extend into the channel. As the channel length is reduced, if the doping is kept constant, the separation between the depletion region boundaries decreases. An increase in the reverse bias across the junctions (with increase in V_{DS}) also pushes the junctions nearer to each other. When the combination of channel length and reverse bias leads to the merging of the depletion regions, punchthrough is said to have occurred.

III REVIEW THEORETICAL CONCEPT [33]

Expressions above for the total power dissipation of CMOS logic gates suggest that we have several different means for reducing the power consumption. These measures include (i) reduction of the power supply voltage V_{DD}, (ii) reduction of the voltage swing in all nodes, (iii) reduction of the switching probability (transition factor) and (iv) reduction of the load capacitance. Note that the switching power dissipation is also a linear function of the clock frequency, yet simply reducing the frequency would significantly diminish the overall system performance. Thus, the reduction of clock frequency would be a viable option only in cases where the overall throughput of the system can be maintained by other means.

The reduction of power supply voltage is one of the most widely practiced measures for low-power design. While such reduction is usually very effective, several important issues must be addressed so that the system performance is not sacrificed. In particular, we need to consider that reducing the power supply voltage leads to an increase of delay.
Also, the input and output signal levels of a low-voltage circuit or module should be made compatible with the peripheral circuitry, in order to maintain correct signal transmission.

The reduction of switching activity requires a detailed analysis of signal transition probabilities, and implementation of various circuit-level and system-level measures such as logic optimization, use of gated clock signals and prevention of glitches. Finally, the load capacitance can be reduced by using certain circuit design styles and by proper transistor sizing.

A. LOW-POWER DESIGN THROUGH VOLTAGE SCALING [33]

Equation 1 show dynamic power dissipation is proportional to the square of the power supply voltage V_{DD}, hence; reduction of V_{DD} will significantly reduce the power consumption. There are two approaches to reduce V_{DD} without performance degrading.

1) *Pipelining Approach*

Now consider an N-stage pipelined structure for implementing the same one logic function. The logic function F(INPUT) has been partitioned into N successive stages, and a total of (N- 1) register arrays have been introduced, in addition to the original input and output registers, to create the pipeline. All registers are clocked at the original sample rate, f_{CLK}. If all stages of the partitioned function have approximately equal delays then the logic blocks between two successive registers can operate N-times slower while *maintaining the same functional throughput* as before. This implies that the power supply voltage can be reduced to a value of V_{DDnew} to effectively slow down the circuit by a factor of N. Here latency is increase by N times.

2) *Parallel Processing Approach*

Another method for trading off area for lower power dissipation is to use parallelism, or hardware replication. This approach could be useful especially when the logic function to be implemented is not suitable for pipelining. Consider N identical processing elements, each implementing the logic function F(INPUT) in parallel. Assume that the consecutive input vectors arrive at the same rate as in the single stage case examined earlier. The input vectors are routed to all the registers of the N processing blocks. Gated clock signals, each with a clock period of (N T_{CLK}), are used to load each register every N clock cycles. This means that the clock signals to each input register are skewed by T_{CLK}, such that each one of the N consecutive input vectors is loaded into a different input register. Since each input register is clocked at a lower frequency of (F_{CLK}/N), the time allowed computing the function for each input vector is increased by a factor of N. This implies that the power supply voltage can be reduced until the critical path delay equals the, new clock period of (N*T_{CLK}). The outputs of the N processing blocks are multiplexed and sent to an output register which operates at a clock frequency of F_{CLK} ensuring the same data throughput rate as before.
Since the time allowed computing the function for each input vector is increased by a factor of N, the power supply voltage can be reduced to a value of V_{DDnew} to effectively slow down the circuit.

B. *CIRCUIT TECHNIQUES FOR LEAKAGE REDUCTION*

1) Standby Leakage Control Using Transistor Stacks (Self-Reverse Bias): Subthershold leakage current flowing through a stack of series-connected transistors reduces when more than one transistor in the stack is turned off. This effect is known as the stacking effect.

2) Multiple threshold Designs: Multiple-threshold CMOS technologies, which provide both high- and low-threshold transistors in a single chip, can be used to deal with the leakage problem. The high-threshold transistors can suppress the subthreshold leakage current, while the low-threshold transistors are used to achieve high performance. Multiple-threshold voltages can be achieved by the following methods.

a) Multiple channel doping.
b) Multiple oxide CMOS
c) Multiple channel length
d) Multiple body bias.

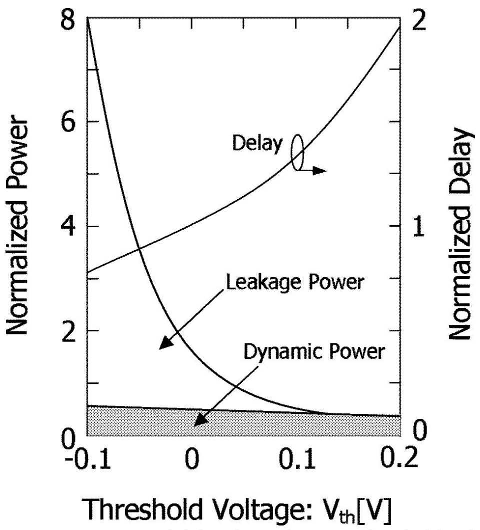

Figure.3 Power and delay dependence on threshold voltage (V_t) [10].

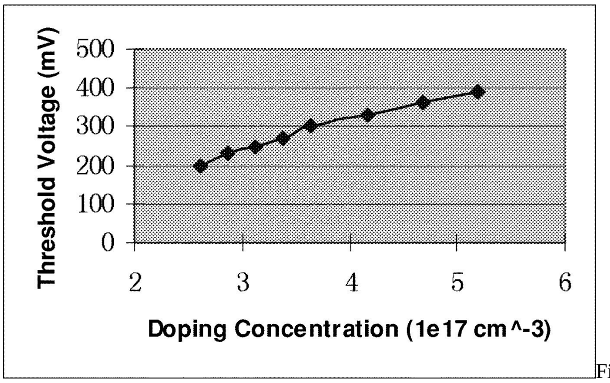

Figure. 4 Vt at different channel-doping densities [19].

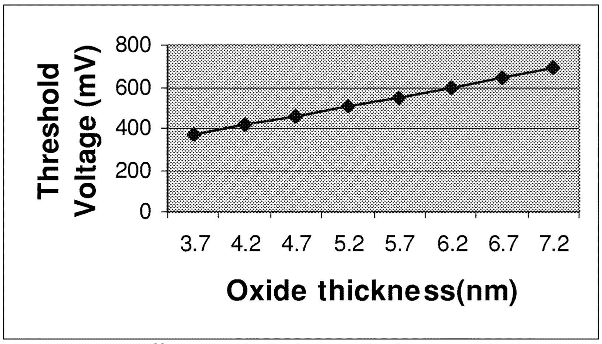

Figure. 5 Vt at different oxide thicknesses [19].

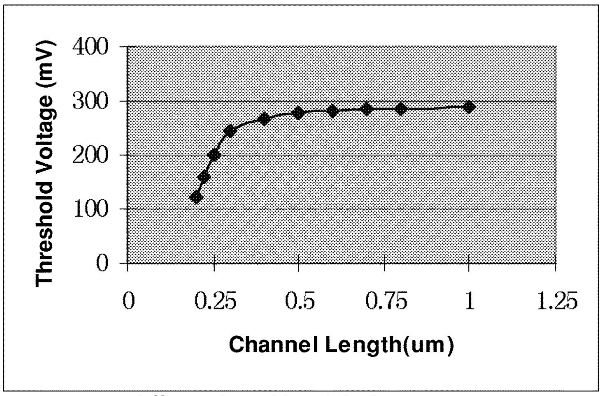

Figure. 6 Vt at different channel length [19].

Multithreshold-voltage CMOS: Multithreshold voltage CMOS (MTCMOS) reduces the leakage by inserting high-threshold devices in series to low V_t circuitry [28]. Figure.7 shows the schematic of an MTCMOS circuit. A sleep control scheme is introduced for efficient power management. In the active mode, SL is set low and sleep control high V_{th} transistors (MP and MN) are turned on. Since their on-resistances are small, the virtual supply voltages (VDDV and VSSV) almost function as real power lines. In the standby mode, SL is set high, MN and MP are turned off, and the leakage current is low.

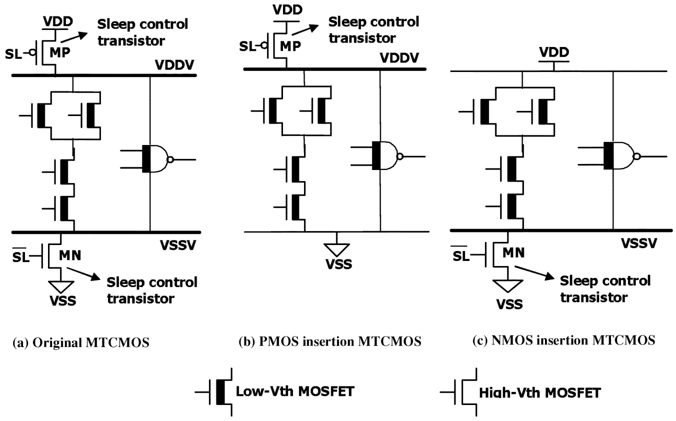

Figure.7 Schematic of MTCMOS circuit [28].

Dual threshold CMOS: For a logic circuit, a higher threshold voltage can be assigned to some transistors in noncritical paths so as to reduce the leakage current, while the performance is maintained due to the use of low threshold transistors in the critical path(s) [15]. Therefore, no additional leakage control transistors are required, and both high performance and low power can be achieved simultaneously.

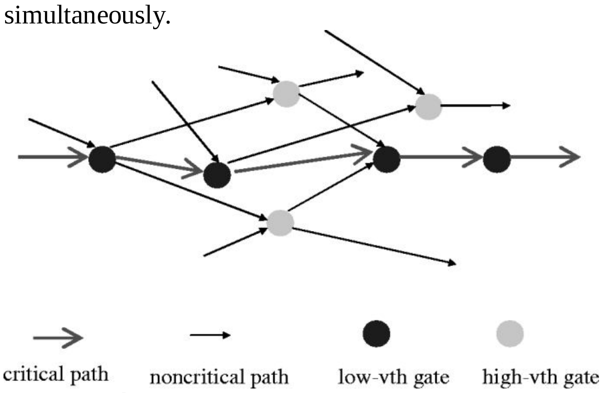

Figure. 8 Dual V_t CMOS circuit.

Variable threshold CMOS: Variable threshold CMOS (VTMOS) is a body-biasing design technique [36]. Figure.9 shows the VTMOS scheme. To achieve different threshold voltages, a self-substrate bias circuit is used to control the body bias. In the active mode, a nearly zero body bias is applied. While in the standby mode, a deeper reverse body bias is applied to increase the threshold voltage and cut off the leakage current.

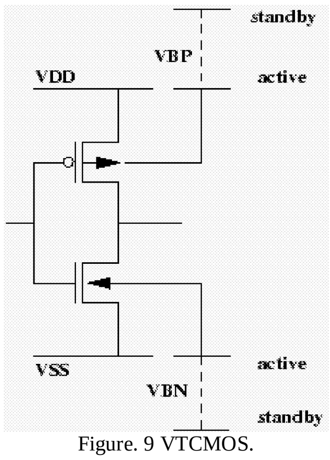

Figure. 9 VTCMOS.

Dynamic threshold CMOS: For dynamic threshold CMOS (DTMOS), the threshold voltage is altered dynamically to suit the operating state of the circuit. A high threshold voltage in the standby mode gives low leakage current, while a low threshold voltage allows for higher current drives in the active mode of operation. Dynamic threshold CMOS can be achieved by tying the gate and body together [7].

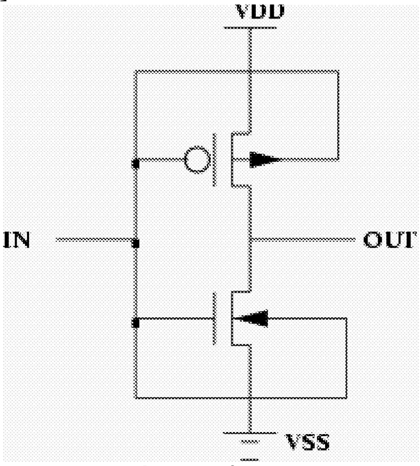

Figure. 10 Schematic of DTMOS inverter.

Double-gate dynamic threshold SOI CMOS (DGDT-MOS): The double-gate dynamic threshold voltage (DGDT) SOI MOSFET [16] combines the advantages of DTMOS and double-gate FD SOI MOSFETs without any limitation on the supply voltage.

3) Dynamic Designs: Dynamic threshold voltage scaling is a technique for active leakage power reduction. This scheme utilizes dynamic adjustment of frequency through back-gate bias control depending on the workload of a system. When the workload decreases, less power is consumed by increasing V_t.

IV LITERATURE REVIEW ON RELATED WORK

[38] Due to the quadratic relationship between dynamic power consumption and V_{DD}, reducing the supply voltage is the most effective way to lower the dynamic power, at the expense of increasing gate delay. In order to prevent the negative effect on performance, the threshold voltage (V_t) must be reduced proportionally with the supply voltage so that a sufficient driving current is maintained. This reduction in the threshold voltage causes an exponential increasing in leakage power, which in turn can raise the static power of the device to unacceptable levels.

To counter the loss in performance while improving the power efficiency, *multiple V_{DD}* [13] and *multiple V_t* [23] techniques have been proposed. The gates on critical paths operate at the higher V_{dd} or lower Vt, while those on non-critical paths operate at the lower V_{dd} or higher V_t, thereby reducing overall power consumption without performance degradation. These techniques have been successfully implemented. For example, IBM's ASIC design flow can fully take advantage of the power-performance trade off by using their voltage island concept and multiple-Vt standard cell library [25]. *Gate sizing* [8] is another powerful method of power optimizing. Logic gates on critical paths may be sized up to meet timing requirement, at the expense of higher power consumption; while those on non-critical paths can be sized down to reduce the power consumption. Hamada *et al.* [18] examined multiple supply voltages, multiple threshold voltages and transistor sizing individually and derived a set of rules of thumb for optimal supply voltages, threshold voltages and transistor sizing. A good summary of these three techniques is presented by Brodersen *et al* [24].

[38] To tackle the ever-increasing leakage power, besides multiple Vt technique, another solution is *stack forcing*. It has been shown that the stacking of two off transistors can significantly reduce leakage power than a single off transistor [27]. The logic gate with stack forcing has much lower leakage power, however, at the expense of a delay penalty, because the effective device width W_{eff} becomes change after stack forcing. It is similar to replacing a low-Vt device with a high-Vt device in a multiple-Vt design.

To achieve the most power efficient design, all these power reduction techniques have to be balanced. [37] combined gate sizing and supply voltage optimization to minimize power consumption under a delay constraint. [12] Presented a heuristic algorithm to combine dual-V_{dd} and dual-Vt techniques. Augsburger [26] evaluated the effectiveness of multiple supply voltage, transistor sizing, and multiple thresholds independently and in conjunction with each other, showing that the order of application of these techniques determines the final savings in active and leakage power.

Recently, researchers have looked at the joint optimization of these techniques, because it can help to achieve maximum power savings compared to a sequential application of a single variable optimization. Sirichotiyakul *et al* [29] presented an algorithm for joint optimization of dual-vt and sizing to reduce leakage power. Karnik *et al* [34] developed a heuristic iterative algorithm to do device sizing and dual-Vt allocation simultaneously to exploit the timing slack for reduction of total power consumption. They found that joint dual-vt and sizing can reduce the power by 10% and 25% compared with pure Vt allocation or pure sizing method, respectively. Srivastava *et al.* [31] were the first to investigate the effectiveness of simultaneously multiple supply and threshold voltage assignment for total power saving. Their algorithm is based on linear programming approach. Nguyen *et al.* [5] developed another linear programming algorithm that can simultaneously perform the threshold voltage assignment and sizing optimization, and then apply the supply voltage optimization as a sequential step. Lee *et al*. [4] proposed heuristic algorithms for simultaneous state, Vt and gate oxide assignment. Srivastava *et al* [32] proposed a sensitivity-based algorithm to perform concurrent sizing, V_{dd} and Vt assignment.

[38] Presented a GA-based power optimization framework that can simultaneously exploit four power optimization techniques: multiple supply voltage assignment, multiple threshold voltage assignment, gate sizing and force stacking.

[35] When the design window is moved toward lower-V_{DD} and lower-V_t along the equi-speed curve, power dissipation is reduced. Since the subthreshold leakage current increases rapidly as V_t is lowered, the power dissipation will be increased again at the point where the leakage current dominates the power dissipation. It can be seen that the power dissipation is at a minimum around where the power dissipation due to the subthreshold leakage current makes up several dozen percentage of the total power dissipation. Lowering both V_{DD} and Vt, however, raises problems. An exponential increase in subthreshold leakage current due to Vt reduction, not only shortens battery life in portable equipment, but also disables the I_{DDQ} testing. For these reasons it is very difficult to lower V_T below 0.2 volts. In addition, significant delay increase due to V_T variation at a low V_{dd}, degrades worst-case circuit speed. However, it is difficult to lower Vt by means of process and device refinement.

[35] There are two approaches to solve these problems. Conventional power-down schemes either on a board or in a chip can solve the battery life problem. The other approach is to control *V,* through substrate bias, which can solve all three problems.

[35] There are three ways to save power dissipation while maintaining maximum operating frequency by utilizing surplus timing in non-critical paths: 1) employing multiple power supplies to lower supply voltage, 2) employing multiple threshold voltages to reduce leakage current, and 3) employing multiple transistor widths, *W's*, to reduce circuit capacitance. In paper derive optimum supply voltages, Optimum threshold voltages and optimum size of transistors for multi supply, multi threshold and multi size transistor designing.

[21] Changing the threshold voltage by altering the gate oxide thickness helps to reduce both subthreshold leakage and gate oxide tunnelling leakage, as well as dynamic power. However, advanced process technology is required for fabricating multiple oxide thickness circuit.

[2] A slight increase in the threshold voltage is shown to have an exponential effect in reducing the total power dissipation. The corresponding increase in the propagation delay is compensated for by increasing the number of buffer stages such that there is still an overall significant reduction in the total power dissipation. As compared to the constant threshold voltage design based on a cost function of 2, the proposed scheme can lead to either a power dissipation reduction of about 70% while maintaining the same delay, or up to 30% in power dissipation with 10% propagation delay reduction, respectively. Closed-form expressions that give the optimum threshold voltage and number of stages are presented.

[3] Level shifters allow for effective interfacing between voltage domains supplied by different voltage levels in multi supply design. In this paper presented a low power level shifters in the 90nm technology node capable of converting subthreshold voltage signals to above threshold voltage signals. The level shifter makes use of MTCMOS design technique which gives more design flexibility, especially in low power systems. The level shifter employs an enable/disable feature, allowing for power saving when the level shifter is not in use.

[17] In this paper, propose two algorithms that apply a Multiple Threshold CMOS (MTCMOS) technique to a single *Vt* (*threshold voltage*) circuit for peak current reduction. Algorithms are more complex than an algorithm of leakage current reduction that uses an MTCMOS technique because current waveform data of a circuit are vectors while leakage current data of a circuit are scalars.

[9] This paper analyzes the potential of fully depleted silicon-on-insulator (FDSOI) technology as a multiple threshold voltage V_T platform for digital circuits compatible with bulk complementary metal–oxide–semiconductor (CMOS). Various technology options, such as gate materials, buried oxide thickness, back plane doping type, and back biasing, were investigated in order to achieve a technology platform that offers at least three distinct V_T options (high-V_T , standard-V_T , and low-V_T). The multi-V_T technology platform highlighted in this paper was developed with standard CMOS circuit design constraints in mind; its compatibility in terms of design and power management techniques, as well as its superior performance with regard to bulk CMOS, are described. Finally, it is shown that a multi-V_T technology platform based on two gate materials offers additional advantages as a competitive solution. The proposed approach enables excellent channel electrostatic control and low V_T variability of the FDSOI process. The viability of the proposed concept has been studied through technology computer-aided design simulations and demonstrated through experimental measurements on 30-nm gate length devices.

V. SCOPE OF RESEARCH

As been mentioned above, and the research papers available have suggested that the research in the low power VLSI design region have continues grooving. From 2003 used different optimization algorithm i.e. linear programming and genetic algorithm for low power design in VLSI. Still has not been developed accurate estimation and optimization tool. So, this field have an enormous scope of research. There is still lack of mathematic use in this field to develop universal estimation and optimization tool.

VI. CONCLUSION

After an extensive survey of literature and research papers related to low power VLSI design, it is concluded that the multiple parameter optimization techniques is used for low power CMOS VLSI design. Major power dissipation in CMOS VLSI circuit is because of dynamic power consumption and leakage current. Assigning lower threshold voltage to critical path and higher threshold to non-critical path power dissipation reduce significant without performance degrading. Linear programming based algorithm is used to find non critical path node in circuit.

Genetic algorithm can be used to find optimum combination of different parameters value for minimise power.

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Design and Simulation of ZIGBEE Transmitter Using Verilog

A. Mohammed Mian and Divyabharathi. R

Abstract: **The past several years have witnessed a rapid development in the wireless network area. So far wireless networking has been focused on high-speed and long range applications. Zigbee technology was developed for a Wireless Personal Area Networks (WPAN), aimed at control and military applications with low data rate and low power consumption. Zigbee is a standard defines the set of communication protocols for low-data-rate short-range wireless networking. Zigbee-based wireless devices operate in 868 MHz, 915 MHz, and 2.4 GHz frequency bands. The maximum data rate is 250K bits per second. Zigbee is mainly for battery-powered applications where low data rate, low cost, and long battery life are main requirements. This paper explores Verilog design for various blocks in Zigbee Transmitter architecture for an acknowledgement frame. The word digital has made a dramatic impact on our society. Developments of digital solutions have been possible due to good digital system design and modeling techniques. Further developments have been made and introduced VLSI in order to reduce size of the architecture, to improve speed of operation, improvements in predictability of the circuit behavior. Digital Zigbee Transmitter comprises of Cyclic Redundancy Check, Bit-to-Symbol block, Symbol-to-chip block, Modulator and Pulse shaping block. The work here is to show how we can design Zigbee transmitter with its specifications by using Verilog with less number of slices and Look up tables (LUTs).**

Keywords: Zigbee, CRC, LUTs, occupied slices.

I. INTRODUCTION

Zigbee is a specification for a suite of high level communication protocols using small, low-power digital radios based on an IEEE 802 standard for personal area networks. Zigbee devices are often used in mesh network form to transmit data over longer distances, passing data through intermediate devices to reach more distant ones. This allows Zigbee networks to be formed ad-hoc, with no centralized control or high-power transmitter/receiver able to reach all of the devices. Any Zigbee device can be tasked with running the network.

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Zigbee operates in the industrial, scientific and medical (ISM) radio bands; 868 MHz in Europe, 915 MHz in the USA and Australia, and 2.4 GHz in most jurisdictions worldwide. Data transmission rates vary from 20 to 900Kbits/second. Zigbee builds upon the physical layer and medium access control defined in IEEE standard 802.15.4 (2003 version) for low-rate WPANs. The specification goes on to complete the standard by adding four main components: network layer, application layer, Zigbee device objects (ZDOs) and manufacturer-defined application objects which allow for customization and favor total integration.

Zigbee is not intended to support power line networking but to interface with it at least for smart metering and smart appliance purposes. Because Zigbee nodes can go from sleep to active mode in 30 ms or less, the latency can be low and devices can be responsive, particularly compared to Bluetooth wake-up delays, which are typically around three seconds. Because Zigbee nodes can sleep most of the time, average power consumption can be low, resulting in long battery life.

II. OBJECTIVE OF THE WORK

Zigbee Transmitter can be designed with analog components. Designing an analog transmitter is easier than digital. But in analog design, data transmission will be poor and the components also bigger and more. This will not allow accurate data transmission. In designing with digital, accurate data transmission will be obtained and power supply voltage range will be smaller. One way of designing digital Zigbee transmitter is with the help of Verilog HDL and VHDL through Xilinx.

The objective is to design and to synthesis the Zigbee transmitter using Verilog which will result in lesser numbers of slices and Look-up-Tables (LUTs) utilized and lossless data transmission. With lesser number of components the power utilized shall be reduced.

III. ZIGBEE TRANSMITTER

A. Specifications

Zigbee digital transmitter in 2.4GHz band is designed using Verilog for acknowledgement frame. The PHY layer supports three frequency bands: a 2.45 GHz band with 16 channels, a 915 MHz band with 10 channels and an 868 MHz band with 1 channel. This paper focuses only on 2.45 GHz band which is used worldwide, with the data rate of 250 Kbps. The MAC layer defines two types of nodes: Reduced Function Devices (RFDs) and Full Function Devices (FFDs). RFDs can only act as end-devices and are equipped with sensors or actuators like transducers, light switches and lamps. They may only interact with a single FFD. FFDs are equipped with a full set of MAC layer functions, which enables them to act as a network coordinator or a network end-device.

Table. 1 Specifications

PARAMETER	SPECIFICATION
Data rate	250 Kbps
No. of channels	16
Operating frequency	2.4 GHz
Channel spacing	5 MHz
Spread spectrum	Direct Sequence Spread Spectrum
Chip rate	2 Mega chips per second
Modulation	OQPSK with Half sine Pulse shaping

The IEEE 802.15.4 defines four MAC frame structures: beacon, data, acknowledgement and MAC command frames. The beacon frame is used by a coordinator to transmit beacons. The function of beacons is to synchronize the clock of all the devices within the same network. The data frame is used to transmit data. Meanwhile, the acknowledgment frame is used to confirm successful frame reception. The MAC commands are transmitted using a MAC command frame.

B. Architecture

The acknowledge frame used contains 11 octets (i.e. 88 bits) of physical protocol data unit (PPDU). The binary data from the PPDU packet are inserted into the cyclic redundancy check block to detect errors during transmission. Every four bits of PPDU octet are mapped onto one data symbol, which will take place in bit-to-symbol block. The symbols will be spread into 32-chip PN sequence by utilizing Direct Sequence Spread Spectrum method in Symbol-to-chip block. Then, the chips will be modulated using OQPSK technique.

C. Existing system

The Zigbee digital transmitter is designed for an acknowledgment frame which is shown in Figure based on IEEE 802.15.4 standard. This is the simplest MAC frame format and does not carry any MAC payload. This frame is constructed from MAC header (MHR) and MAC footer (MFR). The frame control field and direct sequence number (DSN) form the MHR. The MFR is composed of 16-bit frame check sequence (FCS). Both MHR and MFR are known as PHY service data unit (PSDU), which becomes the PHY payload. The PHY payload is prefixed with the synchronization header (SHR) comprised of preamble sequence, start of frame delimiter (SFD), and PHY header (PHR). Together with the SHR, PHR and PHY payload form the PHY protocol data unit (PPDU).

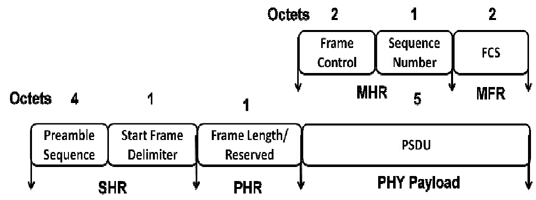


Fig. 1 The Acknowledgement frame

In existing work, the resultant signal from the general architecture was then be amplified and transmitted, which will undergo inter symbol interference. This will result in erroneous information transmission.

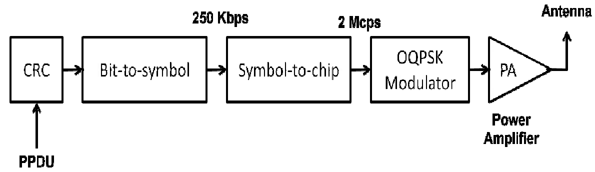


Fig. 2 Transmitter Architecture

E. Proposed system

A Zigbee transmitter is to be designed for PHY and MAC layer for an acknowledgement frame. This design is going to be modeled using Verilog HDL and simulated through Xilinx. The area and performance of operation of the proposed design should satisfy the theoretical specifications and will be verified with the simulation results.

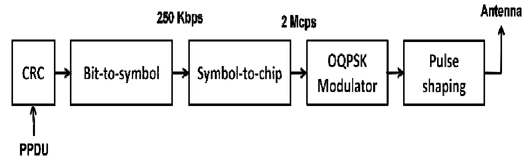


Fig. 3 Proposed Transmitter Architecture

Since the output of the modulator is not assured to be transmitted without error. So, in order to avoid such distortions, we need to add Pulse shaping block at the output of OQPSK modulator. This will avoid Inter symbol Interference and some transmission noises.

IV. DESIGN METHODOLOGY

A. Cyclic redundancy check

Error detection is the process of monitoring data transmission and determining when errors have occurred. Error-detection techniques neither correct errors nor identify which bits are in error – they indicate only when an error has occurred. The purpose of error detection is not to prevent errors from occurring but to prevent undetected errors from occurring.

The most common error-detection techniques are redundancy checking, which includes vertical redundancy checking, checksum, longitudinal redundancy checking, and cyclic redundancy checking.

B. CRC polynomial

The most reliable redundancy checking technique for error detection is a convolutional coding scheme called *cyclic redundancy checking* (CRC). With CRC, approximately 99.999% of all transmission errors are detected. In CRC-16, 16 bits are used for the block check sequence. Here, the entire data stream is treated as a long continuous binary number. Because the Block Check Sequence (BCS) is separate from the message but transported within the same transmission, CRC is considered a *systematic code*. Cyclic block codes are often written as (n, k) cyclic codes where n = bit length of transmission and k = bit length of message. Therefore, the length of the Block Check Character (BCC) in bits is

$$BCC = n - k \tag{1}$$

A CRC-16 BCC is the remainder of a binary division process. A data message polynomial G(x) is divided by a unique generator polynomial function P(x), the quotient is discarded, and the remainder is truncated to 16 bits and appended to the message as a BCS. The generator polynomial must be a prime number. With CRC generation, the division is not accomplished with standard arithmetic division. Instead, modulo-2 division is used, where the remainder is derived from an exclusive OR (XOR) operation. In the receiver, the data stream, including the CRC code, is divided by the same generating function P(x). If no transmission errors have occurred, the remainder will be zero.

Mathematically, CRC can be expressed as

$$\frac{G(x)}{P(x)} = Q(x) + R(x) \tag{2}$$

Where,
G(x) = message polynomial
P(x) = generator polynomial
Q(x) = quotient
R(x) = remainder

The generator polynomial for CRC-16 is

$$P(x) = x^{16} + x^{15} + x^{2} + x^{0} \tag{3}$$

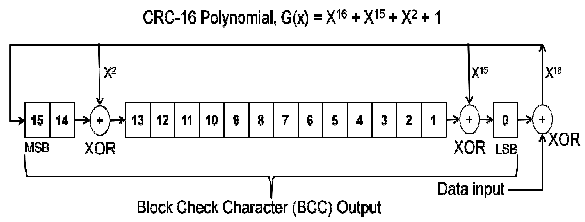


Fig. 4 CRC-16 Generating circuit

A CRC generating circuit requires one shift register for each bit in the BCC. A review of CRC creation process is as follows:

- Get the raw frame
- Left shift the raw frame by n bits and then divide it by P.
- The remainder of the last action is the FCS.
- Append the FCS to the raw frame. The result is the frame to transmit.

CRC-16 detects

- Any single-bit errors
- All double-bit errors
- All odd number of bit errors
- All error bursts of 16 bits or less
- 99.9% of error bursts greater than 16 bits long

C. Bit-to-symbol block

All the 88 bits from the CRC block is inserted into the bit-to-symbol block. This binary information is mapped into the data symbol. The 4 LSBs (b0, b1, b2, b3) of each octet is mapped into one data symbol and the 4 MSBs (b4, b5, b6, b7) of each octet is mapped into the next data symbol. Each octet of PPDU is processed through the bit-to-symbol block sequentially, beginning with the Preamble field and ending with the last octet of the PSDU. For the final result, 22 symbols will be the output of the bit-to-symbol block.

D. Results and discussion

```
module crc(crc16_di, crc16_en, rst, clk, crc16_o);
    input crc16_di;
    input crc16_en;
    input rst;
    input clk;
    output [15:0] crc16_o;
reg crc16_0_in, crc16_2_in, crc16_15_in;
reg [15:0] crc16_o;
```

Fig. 5 CRC Verilog module

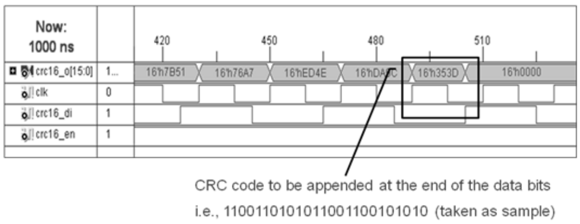


Fig. 6 CRC Simulation Waveform

From the Figure 6.3, the output data consists of 16 digits represented by “crc16_o[15:0]”. At 510ns, with 25 bits of input data, output CRC code is obtained.

```
module bit_sym(clk, rst, data, sym1, sym2, sy
input clk, rst;
input [87:0]data;
output [3:0]sym1, sym2, sym3, sym4, sym5, sym
reg [3:0]sym1, sym2, sym3, sym4, sym5, sym6, :

always@(posedge clk or negedge rst)
begin
if(~rst)
```

Fig. 7 BIT-TO-SYMBOL Verilog module

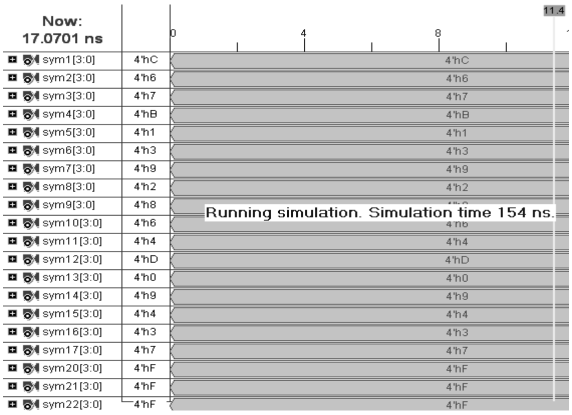


Fig. 8 BIT-TO-SYMBOL Simulation Waveform

```
`timescale 1ns / 1ps
module transmit(data, rst, en, clk, sym1, sym2, sym3, sym4, sym
input data, rst, en, clk;
output [3:0] sym1, sym2, sym3, sym4, sym5, sym6, sym7, sym8, sy
reg [71:0] data_in;
wire [15:0] crc_code;
wire [87:0] crc_out;
always@(clk or data)
   data_in[71:0] = {data_in[70:0],data};
crc block1(data, en, rst, clk, crc_code);
assign crc_out[87:0] = {data_in[71:0], crc_code[15:0]};
bit_sym block2(clk, rst, crc_out, sym1, sym2, sym3, sym4, sym5,

endmodule
```

Fig. 9 Integrated Verilog module

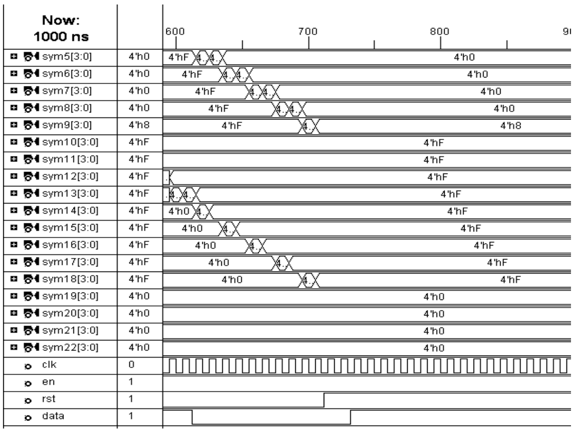
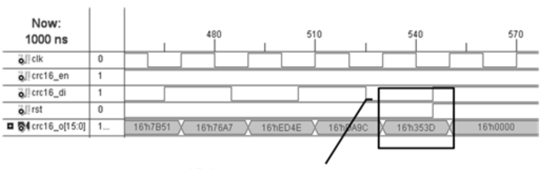


Fig. 10 Simulation Waveform

```
library ieee;
use ieee.std_logic_1164.all;
use ieee.std_logic_arith.all;

entity crc16 is
port( clk:in std_logic;
      crc16_en:in std_logic;
      crc16_di:in std_logic;
      rst:in std_logic;
      crc16_o: out std_logic_vector(15 downto 0));
end entity crc16;
```

Fig. 11 CRC VHDL architecture



BIT-TO-SYMBOL BLOCK

```
library IEEE;
use IEEE.STD_LOGIC_1164.ALL;

entity bit_sym is
    Port ( clk, rst: in  STD_LOGIC;
           data: in std_logic_vector(87 downto 0);
           sym1, sym2, sym3, sym4, sym5, sym6, sym7,
end bit_sym;

architecture behav of bit_sym is
begin
```

Fig. 12 BIT-TO-SYMBOL VHDL architecture

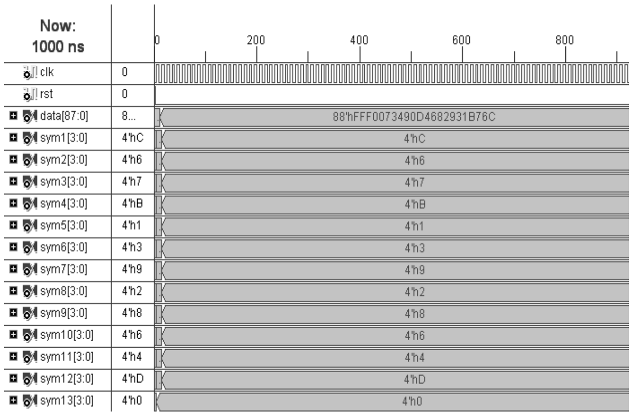


Fig. 13 BIT-TO-SYMBOL Simulation Waveform

```
library IEEE;
use IEEE.STD_LOGIC_1164.ALL;

entity transmit_vh is
    Port ( clk, rst, data : in  STD_LOGIC;
           sym1, sym2, sym3, sym4, sym5, sym6, sym7, sy
end transmit_vh;

architecture Behavioral of transmit_vh is
```

Fig. 14 Transmitter VHDL architecture

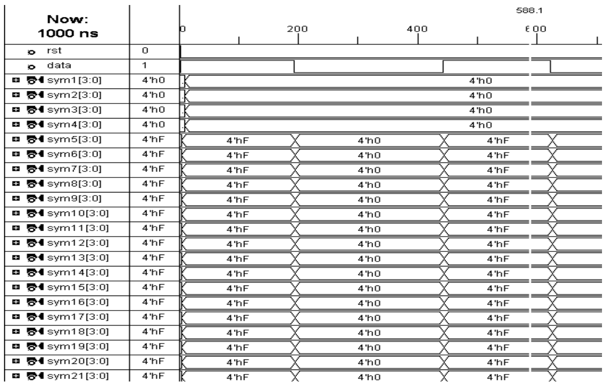


Fig. 15 Simulation Waveform

Device Utilization Summary			
Logic Utilization	**Used**	**Available**	**Utilization**
Number of Slice Flip Flops	16	9,312	1%
Number of 4 input LUTs	3	9,312	1%
Logic Distribution			
Number of occupied Slices	8	4,656	1%
Number of Slices containing only related logic	8	8	100%
Number of Slices containing unrelated logic	0	8	0%
Total Number of 4 input LUTs	3	9,312	1%
Number of bonded IOBs	92	190	48%
IOB Latches	88		
Number of GCLKs	2	24	8%
Total equivalent gate count for design	592		
Additional JTAG gate count for IOBs	4,416		

Fig. 16 Verilog – Device Utilization Summary

Device Utilization Summary			
Logic Utilization	**Used**	**Available**	**Utilization**
Number of Slice Flip Flops	16	9,312	1%
Number of 4 input LUTs	3	9,312	1%
Logic Distribution			
Number of occupied Slices	15	4,656	1%
Number of Slices containing only related logic	15	15	100%
Number of Slices containing unrelated logic	0	15	0%
Total Number of 4 input LUTs	3	9,312	1%
Number of bonded IOBs	91	190	47%
IOB Latches	88		
Number of GCLKs	2	24	8%
Total equivalent gate count for design	592		
Additional JTAG gate count for IOBs	4,368		

Fig.17 VHDL – Device Utilization Summary

The digital transmitter was partially designed and synthesized for Spartan 3E with a speed grade of 5. From the simulation waveform shown before shows the output data for the corresponding inputs. From the Synthesis report shown above, it is obvious that the number of slices utilized using VHDL is more than Verilog HDL.

V. FUTURE WORK PLAN

The Bit-to-Symbol output has to be verified in order to feed the next block. Otherwise this block has been synthesized and number of slices to be utilized has got. To study and design Symbol-to-Chip Mapper. This part has to be designed with the help if Direct Sequence Spread Spectrum Technique. To study and design Offset Quadrature Phase Shift Keying (O-QPSK) modulation technique. This modulation technique alone is suitable for 2.4GHz band of Zigbee transmitter. To study and design Pulse shaping block for modulated output, which will reduce the Inter Symbol Interference (ISI).

VI. CONCLUSION

This paper shows the Verilog based design of digital transmitter for 2.4GHz band Zigbee applications. The behavior of CRC and Bit-to-symbol were characterized using Verilog as well as using VHDL. From the discussion, so far, part of the Zigbee transmitter is alone is characterized and synthesized. Both the results and synthesis report were compared and discussion has made for the design methodology. Thus, using Verilog, Number of occupied slices required is 8 out of 4656 and using VHDL it is 15 out of 4656. The remaining part of the transmitter will be designed and synthesized in future.

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Efficient Localized Broadcasting Using Connected Dominating Sets in Wireless Ad Hoc Networks

Geethu Chandran and Jenopaul.S

Abstract—**Broadcasting, one of the fundamaental operations of the wireless ad-hoc networks, can be implemented using two approaches ie static and dynamic.In broadcasting a node disseminates a message to all other nodes within the network.Usually in static approach the forwarding or non-forwarding status of the node is determined by a globally known priority function and local topology information.The static approach can achieve a constant approximation factor to optimal solution only if position information is available which is not possible in all cases.This paper shows that constant approximation to optimal solution can be obtained using connectivity information only.The status of each node is determined 'on-the-fly' ie while the broadcasting process is being done.This local broadcast algorithm can achieve both full delivery and constant approximation to the optimal solution.The security issues can be solved by comparing the expected and perceived packet delivery ratios.**

Keywords— ***Mobile ad hoc networks, distributed algorithms, broadcasting, connected dominating set, constant approximation***

I. INTRODUCTION

Wireless ad hoc networks are now being used to support wireless networks that can be established without the help of any fixed infrastructure. Wireless devices in an ad hoc networks are usually termed as nodes. One of their important characteristic is their limited transmission ranges. Therefore, each node can directly communicate with only those within its transmission range (i.e., its neighbors) and requires other nodes to act as routers in order to communicate with out-of range destinations. One of the fundamental operations in wireless ad hoc networks is broadcasting, where a node transmits a message to all other where each node on receiving a message transmits nodes in the network. This can be achieved through the traditional process of flooding, it to all its ne ighbors. However, flooding can entail a large number of redundant transmissions, which can lead to significant waste of constrained resources such as bandwidth and power.

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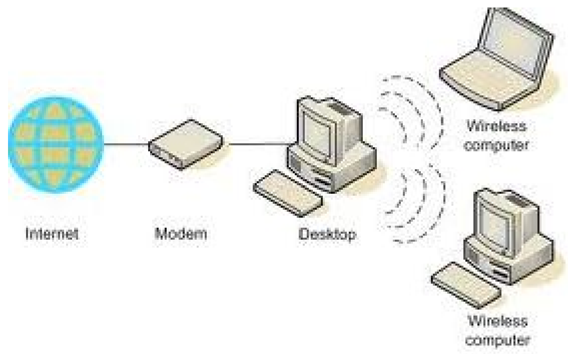

Figure :1-Awireless ad-hoc networks

In general, it is not necessary for every node to forward/transmit the message in order to deliver it to all nodes in the network. A set of nodes form a Dominating Set (DS) if every node in the network is either in the set or has a neighbor in the set. If the nodes in the DS form a connected subgraph then it is called a Connected Dominating Set (CDS). A CDS is hence formed by a source node along with its forwarding nodes. By using only the nodes in the set to forward the message CDS can be used for broadcasting. Therefore, the problems of finding the minimum number of required transmissions (or forwarding nodes) and finding a Minimum Connected Dominating Set (MCDS) can be reduced to each other. Unfortunately, finding a MCDS (and hence minimum number of forwarding nodes) was proven to be NP hard even when the whole network topology is known. A desired objective of many efficient broadcast algorithms is to reduce the total number of transmissions to preferably within a constant factor of its optimum. For local algorithms and in the absence of global network topology information, this is commonly believed to be very difficult or impossible. The existing local broadcast algorithms can be classified based on whether the forwarding nodes are determined statically (based on only local topology information) or dynamically (based on both local topology and broadcast state information). In the static approach, the distinctive feature of local algorithms over other broadcast algorithms is that using local algorithms any local topology changes can affect only the status of those nodes in the neighborhood. Hence, local algorithms can provide scalability as the constructed CDS can be efficiently updated.

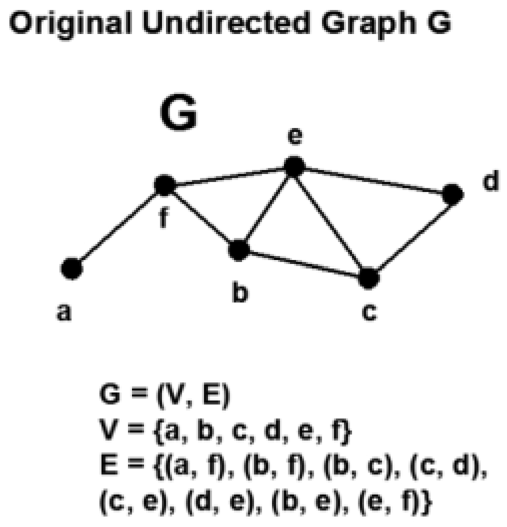

Fig 2:Original undirected graph

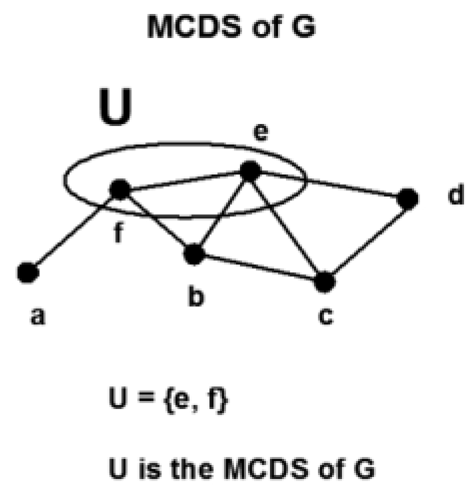

Fig 3:Minimum Connected Dominating Set

The existing local algorithms in this category use a priority function known by all nodes in order to determine the status of each node. Using only local topology information and a globally known priority function, based on the static approach the local broadcast algorithms cannot guarantee a good approximation factor to the optimum solution (i.e., MCDS). On the other hand, in the dynamic approach, the status of each node (hence the CDS) is determined "on-the-fly" during the broadcast progress. Using the dynamic approach, the constructed CDS may vary from one broadcast instance to another even when the whole network topology and the source node remain unchanged. As a result, the broadcast algorithms based on the dynamic approach typically have small maintenance cost and are expected to be robust against node failures and│ changes in network topology.

II. MODEL OF THE NETWORK

We assume that the network consists of a set of nodes $V, |V| = N$. Each node is equipped with omnidirectional antennas. Every node u 2 V has a unique id, denoted id(u), and every packet is stamped by the id of its source node and a nonce, a randomly generated number by the source node. We can assume that all nodes are located in two-dimensional space. However, all the results presented in this paper can be readily extended to three dimensional ad hoc networks. To model the network, we assume two different nodes $u \in V$ and $v \in V$ are connected by an edge if and only if $|uv| \leq R$, where $|uv|$ denotes the Euclidean distance between nodes u and v and R is the transmission range of the nodes. Thus, we can represent the communication graph by G (V,R), where V is the set of nodes and R is the transmission range. This model is, up to scaling, identical to the unit disk graph model, which is a typical model for two dimensional ad hoc networks. Practically speaking, however, the transmission range can be of arbitrary shape as the wireless signal propagation can be affected by many unpredictable factors.

Finally, we assume that the network is connected and static during the broadcast and that there is no loss at the MAC/PHY layer. These assumptions are necessary in order to prove whether or not a broadcast algorithm can guarantee full delivery. Note that without these assumptions even flooding cannot guarantee full delivery.

III. BROADCASTING IN THE DYNAMIC APPROACH

Using the dynamic approach, the status (forwarding/ non forwarding) of each node is determined "on-the-fly" as the broadcasting message propagates in the network. Usually in neighbor-designating broadcast algorithms, each forwarding node selects its own subset of its neighbors to forward the packet and in self-pruning algorithms each node determines its own status based on a self-pruning condition after receiving the first or several copies of the message. It was proved that self-pruning broadcast algorithms are able to guarantee both full delivery and a constant approximation factor to the optimum solution (MCDS) . However, the proposed algorithm in uses position information in order to design a strong self-pruning condition. In the last section, it was observed that position information can simplify the problem of reducing the total number of broadcasting nodes. Moreover, acquiring position information may not be possible in some applications. In this section, we design a hybrid (i.e., both neighbor-designating and self-pruning) broadcast algorithm and show that the algorithm can achieve both full delivery and constant approximation using only the connectivity information.

IV. THE PROPOSED LOCALIZED BROADCAST ALGORITHM

Suppose each node has a list of its 2-hop neighbors (i.e., nodes that are at most 2 hops away). This can be achieved in two rounds of information exchange. In the first round, each node broadcasts its id to its 1-hop neighbors (simply called neighbors). Thus, at the end of the first round, each node has a list of its neighbors. During the second round, each node transmits its id together with the list of its neighbors. The proposed broadcast algorithm is a hybrid algorithm, combining both neighbor designating and self-pruning algorithms and so every node that broadcasts the message may select some of its neighbors to forward the message. In the proposed broadcast algorithm, each broadcasting node selects at most one of its neighbors. A node should broadcast the message if it is selected for forwarding. Other nodes which are not selected have to decide whether or not to broadcast by themselves. This decision is made based on a self-pruning condition called the coverage condition. To evaluate the coverage condition, every node u maintains a list $List^{cov}_u$ (m) for every unique message m. Upon receiving a message m for the first time, $List^{cov}_u$ (m) is created and filled with the ids of all neighbors of u and then updated as follows: Suppose u receives m from its neighbor v and assume that v selects w $\neq$ u to forward the message. Note that w may not be a neighbor of u. However, since w is a neighbor of v, it is at a maximum of 2 hops away from u. Having id's of v and w (included in the message), node u updates $List^{cov}_u$ (m) by removing all nodes in $List^{cov}_u$ (m) that are a neighbor of either v or w. This update can be done because u has a list of its 2-hop neighbors. Since w will eventually broadcast the message, by updating the list, u removes those neighbors that have received the message or will receive it, finally. Every time u receives a copy of message m it updates $List^{cov}_u$(m) as already been explained. If w = u (i.e., u is selected by v to forward the message), node u updates $List^{cov}_u$ (m) by removing only neighbors of v from the list. Note that in this case, u must broadcast the message. However, u has to update $List^{cov}_u$ (m) as it needs to select one of its neighbors from the updated list (if it is not empty) to forward the message.

Definition1 (coverage condition). We say the coverage condition for node u is satisfied at time t if $List^{cov}_u$ (m) =ϕ at time t.

Algorithm 1 shows our proposed hybrid broadcast algorithm. When a node u receives a message m, it creates a list $List^{cov}_u$ (m) if it is not created yet and updates the list as explained earlier. Then, based on whether u was selected to forward or whether the coverage condition is satisfied, u may schedule a broadcast by placing a copy of m in its MAC layer queue. The sources of delay in the MAC layer can be divided into two. Firstly, a message may not be at the head of the queue so it has to wait for other packets to be transmitted. Secondly in contention based channel access mechanisms such as CSMA/CA, to avoid collision, a packet at the head of the queue has to wait for a random amount of time before getting transmitted. In this paper, we assume that a packet can be removed from the MAC layer queue if it is no longer required to be transmitted. Therefore, the broadcast algorithm has access to two functions to manipulate the MAC layer queue. Among the two functions, the first function is the scheduling/placing function, which is used to place a message in the MAC layer queue. We assume that the scheduling function handles duplicate packets, i.e., it does not place the packet in the queue if a copy of it is already in the queue. The second function is used to remove a packet form the queue (it does not do anything if the packet is not in the queue).

Algorithm 1. The proposed hybrid algorithm executed by u

Extract the ids of the broadcasting node and the selected node from the received message m
if u has already broadcast the message m then
Discard the message
Return
end if
if u is receiving m for the first time then
Create and fill the list $List^{cov}_u$ (m)
end if
Update the list $List^{cov}_u$ (m)
Remove the information the previous node had added to message
if $List^{cov}_u$ (m) ; then
Select an id from $List^{cov}_u$ (m) and add it to the message
Schedule the message {(*only update the selected id if m is already in the queue*)}
else {(_$List^{cov}_u$ (m) ; in this case*)}
if u was selected then
Schedule the message
else
Remove the message from the queue if u has not been selected by any node before
end if
end if

The proposed algorithm obeys the following statements:
1. u discards a received message m if it has broadcast m before.
2. If u is selected to forward the message, it schedules a broadcast (regardless of the coverage condition) and never removes the messages from the queue in future. However, u may change or remove the selected node's id from the scheduled message every time it receives a new copy of the message and updates $List^{cov}_u$ (m).
3. Suppose u has not been selected to forward the message by time t and the $List^{cov}_u$ (m) becomes empty at time t after an update. Then at time t, it removes the message from the MAC layer queue (if the message has been scheduled before and is still in the queue).
4. If $List^{cov}_u$ (m) $\neq\phi$ then u selects a node from $List^{cov}_u$ (m) $\neq\phi$ to forward the message and adds the id of the selected node in the message. The selection can be done randomly or based on a criteria. For example, u can select the node with the minimum id or the one with maximum battery life-time.
5. If u has been selected to forward and $List^{cov}_u$ (m) = ϕ it does not select any node to forward the message. This is the only case where a broadcasting node does not select any of its neighbors to forward the message.

V. ANALYSIS OF THE PROPOSED BROADCAST ALGORITHM

In this section, it can be proved that the proposed broadcast algorithm guarantees full delivery as well as a constant approximation to the optimum solution irrespective of the forwarding node selection criteria and the random delay in the MAC layer. In order to prove these properties, assume that nodes are static during the broadcast that the network is connected and there is no loss at the MAC/PHY layer. Note that even flooding cannot guarantee full delivery without these assumptions.

Theorem 5. Algorithm 1 guarantees full delivery.

Proof. Every node broadcasts a message at most once. Therefore, the broadcast process eventually terminates. By contradiction, assume that node d has not received the message by the broadcast termination. Since the network is connected, there is a path from the source nodes (the node that initiates the broadcast) to node d. Clearly, we can find two nodes u and v on this path such that u and v are neighbors, u has received the message and v has not received it. The node u did not broadcast the message since v has not received it. Therefore, u has not been selected to broadcast; thus, the coverage condition must have been satisfied for u. As the result, v must have a neighbor w, which has broadcast the message or was selected to broadcast. Note that all the selected nodes will ultimately broadcast the message. This is a contradiction because, based on the assumption, v should not have a broadcasting neighbor.

Lemma 2. *Using Algorithm 1, the number of broadcasting nodes inside any disk $D_{O,R/2}$ centered at an arbitrary point O and with a radius R/2 is at most 32.*

Proof. All nodes inside $D_{O,R/2}$ are neighbors of each other, thus they receive each others messages. The broadcasting nodes can be divided into two types based on whether or not the coverage condition was satisfied for them just before they broadcast the message. Recall that the coverage condition may be satisfied for a broadcasting node if the node has been selected to forward the message. It is because a selected node has to broadcast the message irrespective of the coverage condition. Consider two disks centered at O with radii R/2 and 3R/2 , respectively. Suppose k is the minimum number such that for every set of k nodes w_i ε $D_{O,3R/2}$, $1 \leq i \leq k$, we have

$$\exists i, j \neq i : |wiwj| \leq R \qquad \text{--------------(1)}$$

Following, we find an upper bound on k. By the minimality of k, there must exist k - 1 nodes w_i ε $D_{O,3R/2}$, $1 \leq i \leq k -1$, such that

$$\forall i, j \neq i : |wiwj| > R \qquad \text{-----------------(2)}$$

Consider k -1 disks $D_1; \ldots ;D_{k-1}$ with radius R/ 2 centered at wi, $1 \leq i \leq k - 1$, respectively. By (2), $D_1,\ldots,D_{k-1}$ are non overlapping disks. Also, every disk D_i, $1 \leq i \leq k - 1$, resides in $D_{O,2R}$ that is the disk centered at O with radius 2R.It is because, the center of every Disk D_i, $1 \leq i \leq k - 1$, is inside $D_{O,3R/2}$. Thus, by an area argument, we get

$$(k-1)(\prod(R/2)^2) \leq \prod(2R)^2 \qquad \text{---------------(3)}$$

Hence, $k \leq 17$.

We first prove that the number of broadcasting nodes inside $D_{O,R/2}$ for which the coverage condition is not satisfied is at most k -1. We then prove the same upper bound for the number of broadcasting nodes inside $D_{O,R/2}$ for which the coverage condition is satisfied. Consequently, the total number of broadcasting nodes inside $D_{O;R/2}$ is bounded by $2k -2 \leq 32$. By contradiction, suppose that there are more than k _ 1 broadcasting nodes inside $D_{O;R2}$ for which the coverage condition is not satisfied. Consider the first k broadcasting nodes be $u_1, \ldots, u_k$ ordered chronologically based on their broadcast time, and $a_1, \ldots, a_k$ the corresponding selected neighbor. Thus, for every i, $1 \leq i \leq k$, we have a_i ε $List^{cov}_{ui}(m)$, where $List^{cov}_{ui}(m)$ is the list of node ui at the time it broadcasts the message. Since $u_1, \ldots, u_k$ are all in $D_{O,R/2}$ and for every i, $1 \leq i \leq k$, $|u_i a_i| \leq R$, we get

$$\forall i, 1 \leq i \leq k : a_i \in D_{O,3R/2} \qquad \text{--------------(4)}$$

Thus, by the definition of k, there are two nodes a_i, a_j,i < j such that $|a_i a_j| \leq R$. The node u_i is broadcast before u_j and is a neighbor of it.Hence, u_j is aware of u_i's selected neighbor a_i and removes a_j from $List^{cov}_{uj}$ (m) as soon as it receives the message from u_i. This is a contradiction because a_j ε $List^{cov}_{uj}(m)$ at the time u_j broadcasts.

It remains to prove that the number of broadcasting nodes inside $D_{O,R/2}$ for which the coverage condition is satisfied is at most k _ 1. By contradiction, suppose that there are at least k broadcasting nodes inside DO;R2 for which the coverage condition is satisfied. Let $v_1, \ldots, v_k$ ε $D_{O,R2}$ be the first k broadcasting nodes, arranged chronologically based on their broadcast time. Note that a broadcasting node must have been selected (by another node) to forward the message if its coverage condition is fulfilled. Let $b_1, b_2, \ldots ,b_k$ be the nodes that selected $v_1, \ldots, v_k$ to forward the message. Therefore, for every i, $1 \leq i \leq k$, we have b_i ε $D_{O,3R/2}$. Also, for every i,$1 \leq i \leq k$ and every j, $1 \leq j \leq k$ and $j \neq i$, we get $b_i \neq b_j$, because each node can select a maximum of one other node to forward. By the definition of k, there must exist two nodes bi and bj, i < j such that $|b_i bj| \leq R$. This is a contradiction because b_i and b_j are neighbors and b_j receives the b_j broadcast message, thus v_j $List^{cov}_{bj}(m)$ as v_i and v_j are neighbors.

Corollary 1. *Let u be any node in the network. Using the proposed Algorithm , the number of broadcasting nodes within the transmission range of u is at most 224.*

Proof. All the nodes within the transmission range of u (including u) are inside a disk with radius R. A disk with radius R can be covered with at most seven disks with radius R/2 . Thus, by Lemma 2, the number of broadcasting nodes within the transmission range of u is at most 7 × 32 = 224.

Theorem 6. *Algorithm 1 has a constant approximation factor to the optimal solution (MCDS). Moreover, the approximation factor is at most 224.*

Proof. Let S_{MCDS} be a MCDS and S_{Alg} be the set of broadcasting nodes using Algorithm 1. Let u be any node in S_{MCDS}. By Corollary 1, the number of broadcasting nodes within the transmission range of u is at most 224. Note that every broadcasting node is within the transmission range of at least one node in S_{MCDS}, because S_{MCDS} is a dominating set. Hence

$$|S_{Alg}| \leq 224 \times |S_{MCDS}|. \quad \text{-----------(5)}$$

VI. IMPLEMENTING STRONG COVERAGE CONDITION

As proven, the proposed broadcast algorithm guarantees that the total number of transmissions is always within a constant factor of the minimum number of required ones. However, the number of transmissions may be further reduced by slightly modifying the broadcast algorithm. As explained earlier, in the proposed algorithm, a selected node has to broadcast the message even if its coverage condition is satisfied. Nevertheless, in some cases, a selected node can avoid broadcasting. For example, a selected node u can abort transmission (by removing the message from the queue) at time t if by time t and based on its collected information, all its neighbors have received the message. This idea can be implemented as follows:

Suppose, for each unique message m, every node u maintains and updates an extra list $List^{str}_u$ (m). Similar to $List^{cov}_u$ (m), $List^{str}_u$ (m) is created and filled with the ids of u's neighbors upon the first reception of message m. Also, every time u receives m, it updates $List^{str}_u$ (m) as follows: Let v be the broadcasting node and $w \neq u$ the selected node by v. Node u first removes the nodes in $List^{str}_u$ (m) that are neighbors of v. If the priority of w (e.g., its id) is higher than u, it also removes the nodes in $List^{str}_u$ (m) that are neighbors of w. To further reduce the number of redundant transmissions, a selected node can abort broadcasting m under the following strong coverage condition.

Definition 2 (strong coverage condition). *It can be said that the strong coverage condition is satisfied for node u at time t if* $List^{str}_u (m) = \phi$ *at time t.*

Note that the strong coverage condition is only used by selected nodes to check whether they need to broadcast. Other nodes make a decision based on the previously defined coverage condition (a weaker condition). The following theorem states that the full delivery is guaranteed if the selected nodes abort transmissions when the strong coverage condition is satisfied. Using a similar approach to that used in the proof of Lemma 2, it can be proven that this extension of the algorithm also achieves a constant approximation factor.

Theorem 7. Suppose Alg-str is a modified version of Algorithm 1 in which each node maintains two lists $List^{cov}_u$ (m) and $List^{str}_u$ (m) and selected nodes can avoid broadcasting under the strong coverage condition. Full delivery can be guaranteed using Alg-str.

VII.SECURITY IN WIRELESS AD-HOC NETWORKS

The wireless ad-hoc networks are easily prone to attacks from malicious nodes that can result in loss of information.The expected and the perceived packet delivery ratios can be compared and in case of abnormalities we can check for the presence of malicious nodes.If the perceived packet delivery ratio is lesser than the expected ratio then we can assume that the packets are being lost.

VIII. EXPERIMENTAL RESULTS

One of the major contributions of this work is the design of a local broadcast algorithm based on the dynamic approach (Algorithm 1) that can achieve both full delivery and a constant approximation factor to the optimum solution without using position information. The simulation experiment is done by distributing the nodes in a square of size of 1,000 ×1,000 m^2. The transmission range is set to 250 m and number of nodes to50.When the simulation begins hello messages are exchanged between the nodes.Then the broadcasting is initiated by a random node after waiting for a stipulated period of time.The x-axis of the

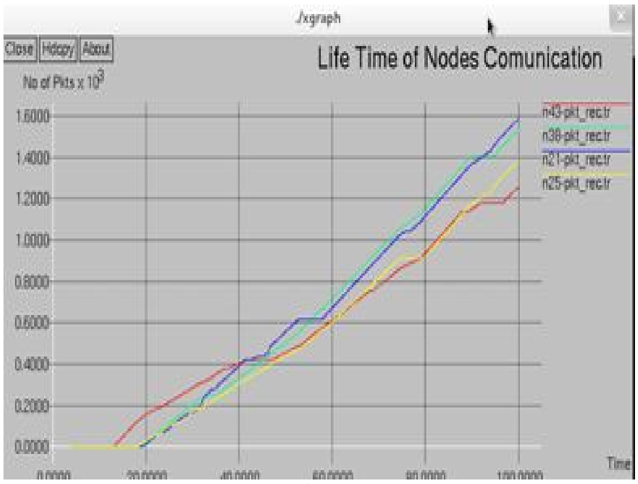

Fig 4:Number of packets versus time

graph shows the number of packets transmitted while y-axis gives the time taken.The number of packets increases drastically after formation of connected dominating sets.Figure 5 shows an instance of broadcasting.The green dots show the various nodes that are receiving the broadcasted packets after the formation of the connected set.

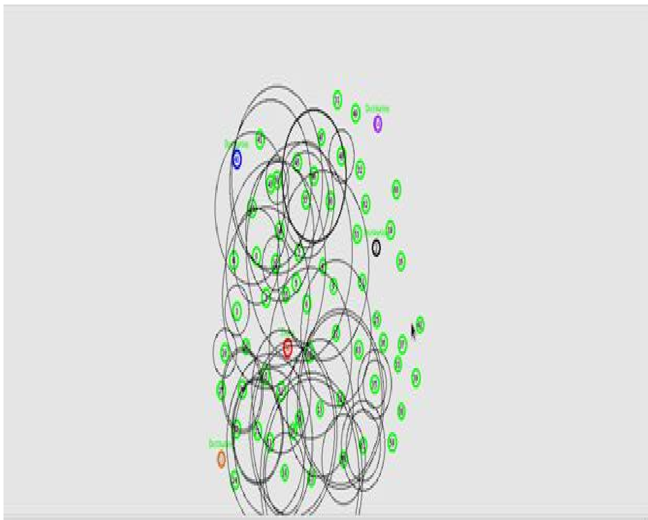

Fig 5:An instance of using the broadcast algorithm

VIII. CONCLUSIONS

In this paper, the capabilities of local broadcast algorithms in reducing the total number of transmissions that are required to achieve full deliverywas investigated. As proven, local broadcast algorithms based on the static approach cannot guarantee a small sized CDS if the position information is not available. It was shown that having relative position information can greatly simplify the problem of reducing the total number of selected nodes using the static approach. In fact, it can be shown that a constant approximation factor is achievable using position information. But by using the dynamic approach, it was shown that a constant approximation is possible using (approximate) position information. This paper shows that local broadcast algorithms that are based on the dynamic approach do not require position information to guarantee a constant approximation factor.

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Comparison of IEEE-754 Standard Single Precision Floating Point Multiplier's

Shaifali and Ms.Sakshi

Abstract: An architecture for a fast 32-bit floating point multiplier using Array Multiplier and Radix-8 Booth's Recoding Algorithm with the single precision IEEE 754-2008 standard has been presented in this paper. Verilog is used to implement a technology-independent design. Floating Point Multiplier is synthesized and targeted for Xilinx Spartan-3E FPGA.

Keywords- Floating point, IEEE-754 Standard, Multiplication, Synthesis, Verilog, Array Multiplier, Modified Booth's Algorithm, Design Compiler, Xilinx

I. INTRODUCTION

Multipliers are key components of many high performance systems such as FIR filters, microprocessors, digital signal processors, etc. Multiplication based operations such as multiply and accumulate(MAC) and inner product are among some of the frequently used computation- intensive arithmetic functions currently implemented in many digital signal processing (DSP) applications such as convolution, fast fourier transform(FFT), filtering and in microprocessors in its arithmetic and logic unit. Since multiplication dominates the execution time of most DSP algorithms, so there is a need of high speed multiplier.

Floating point is a way to represent numbers and do arithmetic in computing machines, ranging from simple calculators to computers. The term floating point is derived from the fact that there is no fixed number of digits before and after the decimal point; that is, the decimal point can float. In general, floating-point representations are slower than fixed-point representations, but they can handle a larger range of numbers.

Floating point multiplication is much like integer multiplication. Because floating-point numbers are stored in sign-magnitude form, the multiplier needs only to deal with unsigned integer numbers and normalization. Integer multiplication using modified booth's algorithm and carry save adder is one way to increase the speed. Because modified Booth algorithm reduces the number of partial products to be generated and is known as the fastest multiplication algorithm and many researches on the multiplier architectures including array, parallel and pipelined multipliers have been pursued which shows that pipelining is the most widely used technique to reduce the propagation delays of digital circuits.

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In this paper an architecture for a fast floating point multiplier compliant with the single precision IEEE 754 standard has been proposed. To attain a generic design, Verilog hardware description language was used for design entry of the entire multiplier unit as it presents a tremendous productivity improvement for circuit designers and descriptions of large circuits can be written in a relatively compact and concise form[1].

II. IEEE-754 FLOATING POINT REPRESENTATION

Over the years, several different floating-point representations have been used in computers; however, for the last ten years the most commonly encountered representation is that defined by the IEEE Standard for Floating-Point Arithmetic (IEEE 754) [3]. It is a technical standard established by the Institute of Electrical and Electronics Engineers (IEEE) and the most widely used standard for floating-point computation, followed by many hardware and software implementations. Single precision representation occupies 32 bits: a sign bit, 8 bits for exponent and 23 for the mantissa.

The most significant bit starts from the left. The three basic components are the sign, exponent, and mantissa. The storage layout for single-precision is show in figure 1:

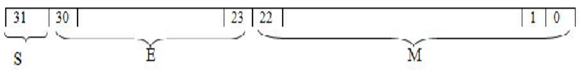

Figure 1: Single Precision Format for Floating Point Numbers[2]

The number represented by the single-precision format is:
Value = $(-1)^s * 2^{E-127} * 1.M$(normalized) when E>0
Where $M = m_{22}\, 2^{-1} + m_{21}\, 2^{-2} + m_{20}\, 2^{-3} + \ldots + m_1\, 2^{-22} + m_0\, 2^{-23}$;
S= sign (0 is positive, 1 is negative)
E =biased exponent; $E_{max} = 255$; $E_{min} = 0$. E=255 and E=0 are used to represent special values.
e =unbiased exponent; e = E-127(bias)

A bias of 127 is added to the actual exponent to make negative exponents possible without using a sign bit. So for example if the value 100 is stored in the exponent placeholder, the exponent is actually -27(100 – 127). In a normalized mantissa, the digit 1 always appears to the left of the decimal point. In fact, the leading 1 is omitted from the mantissa in the IEEE storage format because it is redundant.

Table 1 shows that this format uses 8 bits for exponent with a bias of 127. Twenty-three bits are used as significant (mantissa) with one hidden bit, which will always be concatenated as 1 while being operated.

Table 1: Features of the ANSI/IEEE Standard Floating-Point Representation[4]

Feature	**Single**
Word length, bits	32
Significant bits	23+1(hidden)
Exponent Bits	8
Exponent Bias	127
Zero	(± 0) e + bias = 0, *f* = 0
Denormal	e + bias = 0, *f* ≠0
Infinity	e + bias = 255, *f* = 0
Not-a-Number (NAN)	e + bias = 255, f ≠ 0
Overflow	e+bias >= 255
Underflow	-125 <=e+bias< 0

III. Floating Point Multiplication Algorithm

Normalized floating point numbers have the form of Z= $(-1^{S}) * 2^{(E\ -\ Bias)} * (1.M)$. To multiply two floating point numbers the following is done as shown in figure 2[2]:

a) Multiplying the significand; i.e. (1.M1*1.M2).
b) Placing the decimal point in the result.
c) Adding the exponents; i.e. (E1 + E2 – *Bias*).
d) Obtaining the sign; i.e. s1 xor s2.
e) Normalizing the result; i.e. obtaining 1 at the MSB of the results' significand.
f) Checking for underflow/overflow occurrence.

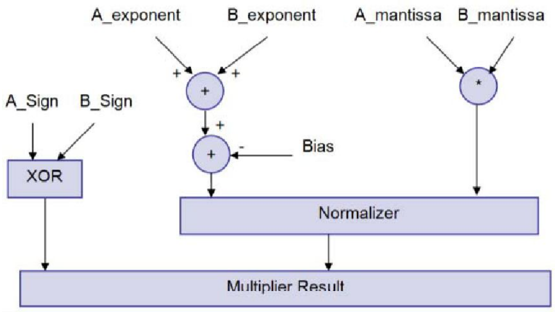

Figure 2: Floating Point Multiplier block diagram[2]

IV. FLOATING POINT MULTIPLIER USING ARRAY MULTIPLIER

This unit is responsible for multiplying the unsigned significand and placing the decimal point in the multiplication product. The result of significand multiplication will be called the intermediate product (IP). The unsigned significand multiplication is done on 24 bit. Multiplier performance should be taken into consideration so as not to affect the whole multiplier's performance. A 24x24 bit carry save multiplier architecture is used as it has a moderate speed with a simple architecture. In the carry save multiplier, the carry bits are passed diagonally downwards (i.e. the carry bit is propagated to the next stage). Partial products are made by ANDing the inputs together and passing them to the appropriate adder. Carry save multiplier has three main stages:

1) The first stage is an array of half adders.
2) The middle stages are arrays of full adders. The number of middle stages is equal to the significand size minus two.
3) The last stage is an array of ripple carry adders. This stage is called the vector merging stage.

The number of adders (Half adders and Full adders) in each stage is equal to the significand size minus one. For example, a 4x4 carry save multiplier is shown in Fig. 5.2 and it has the following stages:

1) The first stage consists of three half adders.
2) Two middle stages; each consists of three full adders.
3) The vector merging stage consists of one half adder and two full adders.

The decimal point is between bits 45 and 46 in the significand multiplier result. The multiplication time taken by the carry save multiplier is determined by its critical path. The critical path starts at the AND gate of the first partial products (i.e. a_1b_0 and a_0b_1), passes through the carry logic of the first half adder and the carry logic of the first full adder of the middle stages, then passes through all the vector merging adders. The critical path is marked in bold in Figure 3.

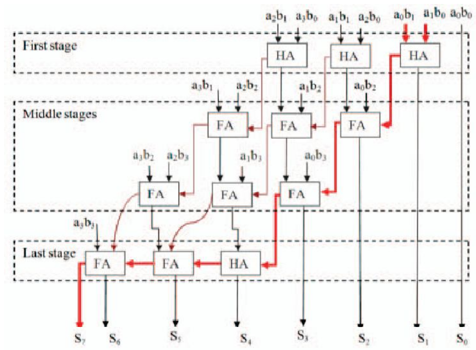

Figure 3: 4x4 bit Carry Save multiplier[2]

Synthesis Results of Floating Point Multiplication using Array Multiplier

a) Synthesis Results on Xilinx

Table 2 shows synthesis results of Floating Point Multiplication using array multiplier on Xilinx.

Table 2: Synthesis Report of Floating Point Multiplier

	Spartan 3E xc3s500E
No. of Slices	773/4656 (16%)
No. of LUTs	1508/9312 (16%)
Minimum Period	31.855ns
Maximum Frequency	31.392MHz

b) Synthesis Results on Design Compiler

Table 3 shows synthesis results of Floating Point Multiplication using array multiplier on Design Compiler.

Table 3: DC synthesis result for multiplier (1 unit = 1 NAND Gate)

Combinational Area	53488.2304690 units
Noncombinational area	4799.691406 units
Total cell area	58286.894531 units
Total Dynamic Power	23.9935 mW

Cell Leakage Power	19.3128 nW

V. FLOATING POINT MULTIPLIER USING RADIX-8 BOOTH'S RECODING ALGORITHM

This is the most important stage, product of the mantissa bits is calculated. The multiplication of mantissa bits is performed in the following stages.

A. Generation of Partial Products

The Booth multiplier makes use of Booth encoding algorithm in order to reduce the number of partial products by considering certain number of bits of the multiplier at a time, thereby achieving a speed advantage over other multiplier architectures. This algorithm is valid for both signed and unsigned numbers. It can handle signed binary multiplication by using 2's complement representation.

For generating the partial products Radix-8 Modified Booth's Algorithm is used. Since the multiplier and multiplicand comprises of 24 bits, this algorithm will generate 8 partial products.

The shortcoming of Radix 2 Booth algorithm is that it becomes inefficient when there are isolated 1's. For example, 001010101(decimal 85) gets reduced to 01-11-11-11-1(decimal 85), requiring eight instead of four operations.001010101(0) recoded as 011111111, requiring 8 instead of 4 operations.

Radix-8 Modified Booth's Algorithm

Recoding extended to 3 bits at a time - overlapping groups of 4 bits each. Radix-8 recoding applies the same algorithm as radix-4, but now we take quartets of bits instead of triplets. Consequently, a multiplier based on this radix-8 scheme generates fewer partial products than a radix-4 multiplier, but the computation of each partial product is more complex[8]. In particular, a partial product corresponding to an encoding x=$\pm$3 requires the computation of 3x, and therefore a full addition. Each quartet is codified as a signed-digit using the table 4:

Table 4: Recoding in Booth Radix-8 Algorithm [8]

Quartet value	**Signed-digit value**	**Quartet value**	**Signed-digit value**
0000	0	1000	-4
0001	+1	1001	-3
0010	+1	1010	-3
0011	+2	1011	-2
0100	+2	1100	-2
0101	+3	1101	-1
0110	+3	1110	-1
0111	+4	1111	0

B. Partial Product Reduction.

8 Partial products are generated using Radix-8 Modified Booth's Algorithm. They are reduced using 4:2 compressors.

Carry Save Adder

A Carry-Save Adder is just a set of one-bit full adders, without any carry-chaining. The most important application of a carry-save adder is to calculate the partial products in integer multiplication. 4:2 compressors are used as carry save adders. The 4:2 compressor structure actually compresses five partial products bits into three. The architecture is connected in such a way that four of the inputs are coming from the same bit position of the weight j while one bit is fed from the neighboring position j-1(known as carry-in). The outputs of 4:2 compressor consists of one bit in the position j and two bits in the position j+1.

A 4:2 compressor can also be built using 3:2 compressors. It consists of two 3:2 compressors (full adders) in series and involves a critical path of 4 XOR delays as shown in Figure 4[8]. The output Cout, being independent of the input Cin accelerates the carry save summation of the partial products. 4:2 compressor is made from 2 full adders. The final carry is saved and hence is called carry save adder. The delay of 4:2 compressor is equal that of 4 xor gates.

Initially two 4:2 compressors are used to reduce each 4 partial products pair to generate the pair of sum and carry. Then these final 4 partial products generated from above two 4:2 compressors are further reduced to generate final sum and carry. The final sum and carry are added in next Carry Propagate adder.

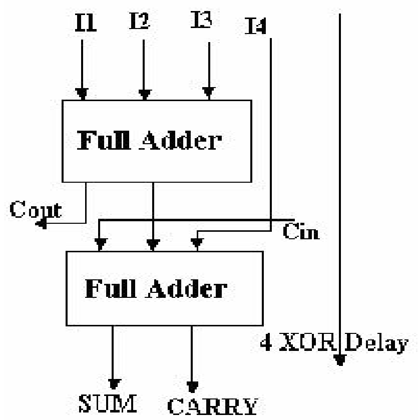

Figure 4: 4:2 Compressor Design using Full Adders

C. Final stage Carry Propagate Adder

Further the partial products generated through carry save adders are further reduced by using Ripple Carry Adder.

Ripple Carry Adder

Ripple Carry Adder is used to obtain the final sum and the output carry by adding the partial products from the carry save adders. It creates a logical circuit using multiple full adders to add N-bit numbers. Each full adder inputs a C_{in}, which is the C_{out} of the previous adder. This kind of adder is a ripple carry adder, since each carry bit "ripples" to the next full adder. The 48-bit sum and carry outputs obtained from the partial product accumulator are added in the final stage adder to give the product of the mantissas.

As shown in Figure 5 a ripple carry adder is a chain of cascaded full adders and one half adder; each full adder has three inputs (A, B, Ci) and two outputs (S, Co). The carry out (Co) of each adder is fed to the next full adder (i.e each carry bit "ripples" to the next full adder).

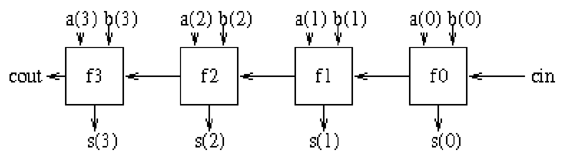

Figure 5: Ripple Carry Adder[2]

Synthesis Results of Unpipelined Floating Point Multiplier

a) Synthesis Results on Xilinx

Table 5 shows the results of Floating point multiplier using Radix-8 Booth's Recoding Algorithm on Xilinx.

Table 5: Synthesis Report of Floating Point Multiplier

	Spartan 3E xc3s500E
No. of Slices	1996 /4656 (42%)
No. of slice FF	185 /9312 (1%)
No. of 4 input LUTs	4059 /9312 (43%)
Minimum Period	22.282ns
Maximum Frequency	44.879MHz

b) Synthesis Results on Design Compiler

Table 6 shows the results of Floating point multiplier using Radix-8 Booth's Recoding Algorithm on Design Compiler.

Table 6: DC synthesis result for Floating Point Multiplier.
(1 unit = 1 NAND Gate)

Combinational Area	169183.312500 units
Noncombinational area	11902.347656 units
Total cell area	181079.031250 units
Total Dynamic Power	54.4868 mW
Cell Leakage Power	60.3955 nW

VI. NORMALIZATION

The result of the significand multiplication (intermediate product) must be normalized to have a leading '1' just to the left of the decimal point (i.e. in the bit 46 in the intermediate product). Since the inputs are normalized numbers then the intermediate product has the leading one at bit 46 or 47

a) If the leading one is at bit 46 (i.e. to the left of the decimal point) then the intermediate product is already a normalized number and no shift is needed.

b) If the leading one is at bit 47 then the intermediate product is shifted to the right and the exponent is incremented by 1[2].

VII. CONCLUSION

A hardware implementation of a high speed floating point multiplier based on the IEEE-754 single precision format is developed using Array multiplier and Radix-8 booth's recoding algorithm. The modules are written in Verilog HDL to optimize implementation on FPGA. After implementation on a Xilinx Spartan 3E FPGA it achieves 31.392MHz using Array multiplier and 44.879MHz using Radix-8 booth's recoding algorithm.

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A Survey on Recent Trends in Wireless Sensor Networks

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***ABSTRACT------*Wireless Sensor Networks have promise a large variety of applications. They are often deployed in potentially adverse or even hostile environments. Intrusion detection systems make available a necessary layer of in-depth fortification for wired networks. Miniature research has been performed about intrusion detection in the areas of wireless sensor networks. Energy efficiency in wireless sensor network [WSN] is the highly important role for the researchers. Clustering is the important factors for real time applications .We present the challenge of constructing intrusion detection systems for wireless sensor networks and survey the intrusion detection techniques, and indicate important future research directions.**

Keywords: Clustering, Issues of WSN, Attacks, Load Balancing, Sensor Technology

I. INTRODUCTION

A wireless sensor node (WSN) is a one type of sensor technology to monitor physical or conservational needs, such as pressure, sound, vibration, temperature, motion and to transmit the data to a sink (base station) through the network. Currently most of the latest networks are bi-directional, enabling to cope up with the activity of the sensors [1]. Military applications like battle field reconnaissance is the main inspiration for wireless sensor networks development, recently this type of distributed networks a read opted in most of remote monitoring applications and industrial measurements application like machine condition monitoring, industrial process monitoring, structural health monitoring, and indoor monitoring.Sensors nodes are characteristically proficient of wireless communication and are considerably obliged in the amount of existing resources such as energy (power), storage (memory) and computation [2]. These obligate make the deployment and operation of WSN significantly distinct from existing wireless networks, and demand the development of resource aware protocols and supervision techniques.

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II. CLUSTERING APPROACH

Clustering is considered as an effective approach to reduce network overhead and improve scalability. Since sensor networks are based on the dense deployment of disposable and low-cost sensor nodes, destruction of some nodes by hostile action does not affect a military operation as much as the destruction of a traditional sensor, which makes the sensor network concept a better approach for battlefields [3]. The transmission between the two nodes will minimize the other nodes to show the improve throughput and greater than spatial reuse than wireless networks to lack the power controls. Adaptive Transmission Power technique to improve the Network Life Time in Wireless Sensor Networks.

The Clustering Technique using the minimum spanning tree[MST] to detect the shortest path in wireless sensor networks. The data from nearby the cluster heads will be directly transmitted to the sink node. The data from sink nodes to calculate the distance whereas the cluster head will be transmitted through the shortest multihop path. The distance between the cluster head and sink node. The shortest path between each cluster head to the sink node. To find the Predominant node[Maximum number of path].Transmission power techniques is to improve the performance of the network in several aspects [4]. Transmission range in the wireless networks should be change the ranges in each link. The traffic capacity decreases when more nodes are added to increases the interference. Routing graph theory to multiple paths from data sources to a neighbor's node.

In the cluster-based approach sensor nodes in particular WSN are permitted to transmit sensed data towards the base station. In this allows sensor nodes to sense and transmit the sensed information to the cluster-heads directly, instead of routing through its immediate neighbors. When a cluster node fails because of energy depletion we need to choose alternative cluster for that particular region. In periodical time each sensor node in the cluster should possess the next cluster head re-election based on energy to avoid node failure. Unlike previous algorithms, cluster formation precedes before cluster head selection. The spanning tree is 'minimal' to the cluster of each node when the total length of the edges is the minimum necessary to connect all the vertices in the clustering head.

In the newly formed clusters, the each node with the highest energy level is selected as the cluster head and the next higher energy level node is selected as the next CH node.

III. ROUTING PROTOCOL

DTN is fundamentally an opportunistic communication system, where communication links only exist temporarily, rendering it impossible to establish end-to-end connections for data delivery. In such networks, routing is largely based on nodal contact probabilities. The key design issue is how to efficiently maintain, update, and utilize such probabilities. Clustering is considered as an effective approach to reduce network overhead and improve scalability. Various clustering algorithms have been investigated in the context of mobile ad hoc networks. However, none of them can be applied directly to DTN, because they are designed for well-connected networks and require timely information sharing among nodes [5]. A node in real-life tends to visit some locations more frequently than others. If two nodes share the same home location, they have high chance to meet each other. Thus real-life mobility patterns naturally group mobile devices into clusters.

Due to possible errors in the estimation of contact probabilities and unpredictable sequence of the meetings among mobile nodes, many unexpected small size clusters may be formed. To deal with this problem, we employ a merging process that allows a node to join a “better” cluster, where the node has a higher stability as to be discussed in the next section. The merging process is effective to avoid fractional clusters.

IV. LOAD BALANCING

Load balancing is an effective enhancement to the proposed routing protocol. The basic idea is to share traffic load among cluster members in order to reduce the dropping probability due to queue overflow at some nodes. Sharing traffic inside a cluster is reasonable, because nodes in the same cluster have similar mobility pattern, and thus similar ability to deliver data messages [6]. More specifically, it randomly transmits as many messages as possible to any node it meets, until their queues are equally long .

V. CATEGORIES OF SENSOR NODES

5.1 Passive, Omni Directional Sensors: passive sensor nodes sense the environment without manipulating it by active probing. In this case, the energy is needed only to amplify their analog signals. There is no notion of “direction” in measuring the environment.

5.2 Passive, narrow-beam sensors: these sensors are passive and they are concerned about the direction when sensing the environment.

5.3 Active Sensors: these sensors actively probe the environment.

Since a sensor node has limited sensing and computation capacities, communication performance and power, a large number of sensor devices are distributed over an area of interest for collecting information (temperature, humidity, motion detection, etc.). These nodes can communicate with each other for sending or getting information either directly or through other intermediate nodes and thus form a network, so each node in a sensor network acts as a router inside the network [7]. In direct communication routing protocols (single hop), each sensor node communicates directly with a control center called Base Station (BS) and sends gathered information. The base station is fixed and located far away from the sensors. Base station(s) can communicate with the end user either directly or through some existing wired network. The topology of the sensor network changes very frequently. Nodes may not have global identification. Since the distance between the sensor nodes and base station in case of direct communication is large, they consume energy quickly.

VI. TRANSMITTED POWER

Wireless sensor networks (WSNs) provide a new class of computer systems and expand human ability to remotely interact with the physical world. Most of the sensors used so far are point sensors which have disc-shaped sensing and communication areas. Energy-efficient communication is discussed in WSNs. Saving energy is very important in WSNs because of the limited power supply of sensors and the inconvenience to recharge their batteries [8]. Methods are proposed to reduce communication energy by minimizing the total sensor transmission power. That is, instead of transmitting using the maximum possible power, sensors can collaboratively determine and adjust their transmission power to reach minimum total transmission power and define the topology of the WSN by the neighbor relation under certain criteria. This is in contrast to the “traditional” network in which each node transmits using its maximum transmission power and the topology is built implicitly without considering the power issue. Choosing the right transmission power critically affects the system performance in several ways. First, it affects network spatial reuse and hence the traffic carrying capacity. Choosing too large power level results in excessive interference, while choosing too small power level results in a disconnected network. Second, it impacts on the contention for the medium. Collisions can be mitigated as much as possible by choosing the smallest transmission power subject to maintaining network connectivity [9]. The goal is to find distributed methods to let each sensor decide its transmission power by communicating with other sensors to minimize total sensor transmission power while maintaining the connectivity of the network. It is pointed out that it can maintain the network connectivity, but may not minimize the total sensor transmission power. Then it is enhanced to DTCYC algorithm, where the basic idea is to let each sensor remove the largest edge in every cycle involving it as a vertex.

6.1 Power Efficiency in WSNs is generally accomplished in three ways:

- Low duty cycle operation
- Local/In network processing to reduce data volume (and hence transmission time)
- Multihop networking reduces the requirement for long range transmission since signal path loss is an inverse exponent with range of distance. Each node in the sensor network can act as a repeater, thereby reducing the link range coverage required and in turn the transmission power.

VII. ADVANTAGES AND DISADVANTAGES

7.1 Advantages

- Network setups can be done without fixed infrastructure.
- Ideal for the non-reachable places such as across the sea, mountains, rural areas or deep forests.
- Flexible if there is ad hoc situation when additional workstation is required.
- Implementation cost is cheap.

7.2 Disadvantages

- Less secure because hackers can enter the access point and get all the information.
- Lower speed compared to a wired network.
- More complex to configure than a wired network.

VIII. OVERVIEW OF SENSOR TECHNOLOGY

Sensor Nodes are almost invariably constrained in energy supply and radio channel transmission bandwidth, these constraints, in conjunction with a typical deployment of large number of sensor nodes, have posed a plethora of challenges to the design and management of WSNs [9]. Some of the key technologies and standards elements that are relevant to sensor networks are as follows:

8.1 Sensors

- Intrinsic Functionality
- Signal processing
- Compression, forward error correction, encryption
- Control/actuation
- Clustering and in-network computation
- Self assembly

8.2 Wireless Radio Technologies

- Software defined radios
- Transmission range
- Transmission impairments
- Modulation Techniques
- Network Topologies

8.3 Standards

- IEEE 802.1.1a/b/g together with ancillary security protocols
- IEEE 802.15.1 PAN/Bluetooth
- IEEE 802.15.3 Ultra wide band (UWB)
- IEEE 802.15.4 ZIGBEE
- IEEE 802.16 WIMAX
- IEEE 1451.5 (Wireless Sensor Working Group)
- Mobile IP

8.4 Software Applications

- Operating Systems
- Network Software
- Direct database Connectivity software
- Middleware software
- Data Management Software

IX. ISSUES OF WIRELESS SENSOR NETWORKS

9.1 Hardware and Operating System for WSN

Wireless sensor networks are composed of hundreds of thousands of tiny devices called nodes. A sensor node is often abbreviated as a node. A Sensor is a device which senses the information and passes the same on to a mote. Sensors are used to measure the changes to physical environment like pressure, humidity, sound, vibration and changes to the health of person like blood pressure, stress and heart beat [10]. A Mote consists of processor, memory, battery, A/D converter for connecting to a sensor and a radio transmitter for forming an ad hoc network. A Mote and Sensor together form a Sensor Node. There can be different Sensors for different purposes mounted on a Mote. Motes are also sometimes referred to as Smart Dust. A Sensor Node forms a basic unit of the sensor network

9.2. Wireless Radio Communication Characteristics

Performance of wireless sensor networks depends on the quality of wireless communication. But wireless communication in sensor networks is known for its unpredictable nature. Main design issues for communication in WSNs are:

- Low power consumption in sensor networks is needed to enable long operating lifetime by facilitating low duty cycle operation and local signal processing.
- Distributed sensing effectively acts against various environmental obstacles and care should be taken that the signal strength, consequently the effective radio range is not reduced by various factors like reflection, scattering and dispersions.
- Multihop networking may be adapted among sensor nodes to reduce the range of communication link.
- Long range communication is typically point to point and requires high transmission power, with the danger of being eavesdropped. So, short range transmission should be considered to minimize the possibility of being eavesdropped.
- Communication systems should include error control subsystems to detect errors and to correct them.

9.3. Deployment

Deployment means setting up an operational sensor network in a real world environment [11]. Deployment of sensor network is a labor intensive and cumbersome activity as it does not have influence over the quality of wireless communication and also the real world puts strains on sensor nodes by interfering during communications. Sensor nodes can be deployed either by placing one after another in a sensor field or by dropping it from a plane.

9.4 Localization

Sensor localization is a fundamental and crucial issue for network management and operation. In many of the real world scenarios, the sensors are deployed without knowing their positions in advance and also there is no supporting infrastructure available to locate and manage them once they are deployed. Determining the physical location of the sensors after they have been deployed is known as the problem of localization.

9.5 Synchronization

Clock synchronization is an important service in sensor networks. Time Synchronization in a sensor network aims to provide a common timescale for local clocks of nodes in the network. A global clock in a sensor system will help process and analyze the data correctly and predict future system behavior [12]. Some applications that require global clock synchronization are environment monitoring, navigation guidance, vehicle tracking etc. A clock synchronization service for a sensor network has to meet challenges that are substantially different from those in infrastructure based networks.

9.6 Calibration

Calibration is the process of adjusting the raw sensor readings obtained from the sensors into corrected values by comparing it with some standard values [13]. Manual calibration of sensors in a sensor network is a time consuming and difficult task due to failure of sensor nodes and random noise which makes manual calibration of sensors too expensive.

9.11 Network Layer Issues

Energy efficiency is a very important criterion. Different techniques need to be discovered to eliminate energy inefficiencies that may shorten the lifetime of the network. At the network layer, various methods need to be found out for discovering energy efficient routes and for relaying the data from the sensor nodes to the BS so that the lifetime of a network can be optimized. Routing Protocols should incorporate multi-path design technique. Multi-path is referred to those protocols which set up multiple paths so that a path among them can be used when the primary path fails. Path repair is desired in routing protocols when ever a path break is detected. Fault tolerance is another desirable property for routing protocols. Routing protocols should be able to find a new path at the network layer even if some nodes fail or blocked due to some environmental interference. Sensor networks collect information from the physical environment and are highly data centric [14]. In the network layer in order to maximize energy savings a flexible platform need to be provided for performing routing and data management. The data traffic that is generated will have significant redundancy among individual sensor nodes since multiple sensors may generate same data within the vicinity of a phenomenon. The routing protocol should exploit such redundancy to improve energy and bandwidth utilization. As the nodes are scattered randomly resulting in an ad hoc routing infrastructure, a routing protocol should have the property of multiple wireless hops.

9.12 Quality of Service

Quality of service is the level of service provided by the sensor networks to its users. Quality of Service (QoS) for sensor networks as the optimum number of sensors sending information towards information-collecting sinks or a base station.

9.13 Security

Security in sensor networks is as much an important factor as performance and low energy consumption in many applications. Security in a sensor network is very challenging as WSN is not only being deployed in battlefield applications but also for surveillance, building monitoring, burglar alarms and in critical systems such as airports and hospitals. Since sensor networks are still a developing technology, researchers and developers agree that their efforts should be concentrated in developing and integrating security from the initial phases of sensor applications development; by doing so, they hope to provide a stronger and complete protection against illegal activities and maintain stability of the systems at the same time.

X.ATTACKS ON WIRELESS SENSOR NETWORK

10.1 Introduction

Many sensor network routing protocols are quite simple, and for this reason are sometimes even more susceptible to attacks against general ad-hoc routing protocols.

Attacks can be classified into two major categories, according to the interruption of communication act, namely

1) Passive attacks
2) Active attacks

From this regard, when it is referred to a passive attack it is said that the attack obtain data exchanged in the network without interrupting the communication [15]. When it is referred to an active attack it can be affirmed that the attack implies the disruption of the normal functionality of the network, meaning information interruption, modification, or fabrication.

Examples of passive attacks are eavesdropping, traffic analysis, and traffic monitoring. Examples of active attacks include jamming, impersonating, modification, Denial of Service (DoS), and message replay.

10.2 Traffic analysis

Traffic analysis is the process of intercepting and examining messages in order to deduce information from patterns in communication.

10.3 Denial-of-service attack (DoS attack) or Distributed Denial-of-Service attack (DDoS attack)

A Denial-of- service attack (DoS attack) or distributed denial of service attack (DDoS attack) is an attempt to make a computer resource unavailable to its intended users [16]. Although the means to carry out, motives for, and targets of a DoS attack may vary, it generally consists of the concerted efforts of a person or persons to prevent an Internet site or service from functioning efficiently or at all, temporarily or indefinitely. Perpetrators of DoS attacks typically target sites or services hosted on high profile web servers such as banks, credit card payment gateways, and even root name servers .

10.4 Replay attack

A replay attack is a breach of security in which information is stored without authorization and then retransmitted to trick the receiver into unauthorized operations such as false identification or authentication or a duplicate transaction. For example, messages from an authorized user who is logging into a network may be captured by an attacker and resent (replayed) the next day. Even though the messages may be encrypted, and the attacker may not know what the actual keys and passwords are, the retransmission of valid log on messages is sufficient to gain access to the network [17]. Also known as a "man-in-the-middle attack", a replay attack can be prevented using strong digital signatures that include time stamps and inclusion of unique information from the previous transaction such as the value of a constantly incremented sequence number.

10.5 Eavesdropping

Eavesdropping is the intercepting and reading of messages and conversations by unintended receivers. The mobile hosts in mobile ad hoc networks share a wireless medium. The majorities of wireless communications use the RF spectrum and broadcast by nature. Signals broadcast over airwaves can be easily intercepted with receivers tuned to the proper frequency. Thus, messages transmitted can be overheard, and fake messages can be injected into network.

10.6 Interference and Jamming

Radio signals can be jammed or interfered with, which causes the message to be corrupted or lost. If the attacker has a powerful transmitter, a signal can be generated that will be strong enough to overwhelm the targeted signals and disrupt communications [18]. The most common types of this form of signal jamming are random noise and pulse. Jamming equipment is readily available. In addition, jamming attacks can be mounted from a location remote to the target networks.

10.7 Flooding

Routing message flooding attacks, such as hello flooding, RREQ flooding, acknowledgement flooding, routing table overflow, routing cache poisoning, and routing loop are simple examples of routing attacks targeting the route discovery phase. Proactive routing algorithms, such as DSDV and OLSR, attempt to discover routing information before it is needed, while reactive algorithms, such as DSR and AODV, create routes only when they are needed.

10.8 Data forwarding phase

Some attacks also target data packet forwarding functionality in the network layer. In this scenario the malicious nodes participate cooperatively in the routing protocol routing discovery and maintenance phases, but in the data forwarding phase they do not forward data packets consistently according to the routing table. Malicious nodes simply drop data packets quietly, modify data content, replay, or flood data packets; they can also delay forwarding time-sensitive data packets selectively or inject junk packets.

10.9 Particular routing protocols

There are attacks that target some particular routing protocols. In DSR, the attacker may modify the source route listed in the RREQ or RREP packets. It can delete a node from the list, switch the order, or append a new node into the list. In AODV, the attacker may advertise a route with a smaller instance metric than the actual distance, or advertise a routing update with a large sequence number and invalidate all routing updates from other nodes.

10.10 Wormhole attack

In the wormhole attack, an adversary tunnels messages received in one part of the network over a low latency link and replays them in a different part. The simplest instance of this attack is a single node situated between two other nodes forwarding messages between the two of them [19]. However, wormhole attacks more commonly involve two distant malicious nodes colluding to understate their distance from each other by relaying packets along an out-of-bound channel available only to the attacker. An adversary situated close to a base station may be able to completely disrupt routing by creating a well-placed wormhole. An adversary could convince nodes who would normally be multiple hops from a base station that they are only one or two hops away via the wormhole. This can create a sinkhole: since the adversary on the other side of the wormhole can artificially provide a high-quality route to the base station, potentially all traffic in the surrounding area will be drawn through her if alternate routes are significantly less attractive. This will most likely always be the case when the endpoint of the wormhole is relatively far from a base station. Wormholes can also be used simply to convince two distant nodes that they are neighbors by relaying packets between the two of them. Wormhole attacks would likely be used in combination with selective forwarding or eavesdropping. Detection is potentially difficult when used in conjunction with the Sybil attack.

10.11 Rushing attack

Two colluded attackers use the tunnel procedure to form a wormhole. If a fast transmission path (e.g. a dedicated channel shared by attackers) exists between the two ends of the wormhole, the tunneled packets can propagate faster than those through a normal multi-hop route. This forms the rushing attack. The rushing attack can act as an effective denial of-service attack against all currently proposed on-demand WSN routing protocols, including protocols that were designed to be secure, such as ARAN and Ariadne.

10.12 Resource consumption attack

This is also known as the sleep deprivation attack. An attacker or a compromised node can attempt to consume battery life by requesting excessive route discovery, or by forwarding unnecessary packets to the victim node.

10.13 Location disclosure attack

An attacker reveals information regarding the location of nodes or the structure of the network. It gathers the node location information, such as a route map, and then plans further attack scenarios.

10.14 SYN flooding attack

The SYN flooding attack is a denial-of-service attack. The attacker creates a large number of half-opened TCP connections with a victim node, but never completes the handshake to fully open the connection. For two nodes to communicate using TCP, they must first establish a TCP connection using a three-way handshake. The three messages exchanged during the handshake allow both nodes to learn that the other is ready to communicate and to agree on initial sequence numbers for the conversation. During the attack, a malicious node sends a large amount of SYN packets to a victim node, spoofing the return addresses of the SYN packets. The SYN-ACK packets are sent out from the victim right after it receives the SYN packets from the attacker and then the victim waits for the response of ACK packet [20]. Without receiving the ACK packets, the half-open data structure remains in the victim node. If the victim node stores these half-opened connections in a fixed size table while it awaits the acknowledgement of the three-way handshake, all of these pending connections could overflow the buffer, and the victim node would not be able to accept any other legitimate attempts to open a connection.

10.15 Session hijacking

Session hijacking takes advantage of the fact that most communications are protected (by providing credentials) at session setup, but not thereafter. In the TCP session hijacking attack, the attacker spoofs the victim's IP address, determines the correct sequence number that is expected by the target, and then performs a DoS attack on the victim. Thus the attacker impersonates the victim node and continues the session with the target. Hijacking a session over UDP is the same as over TCP, except that UDP attackers do not have to worry about the overhead of managing sequence numbers and other TCP mechanisms. Since UDP is connectionless, edging into a session without being detected is much easier than the TCP session attacks.

10.16 Malicious code attacks

Malicious code, such as viruses, worms, spywares, and Trojan Horses, can attack both operating systems and user applications. These malicious programs usually can spread themselves through the network and cause the computer system and networks to slow down or even damaged.

10.17 Repudiation attacks

Repudiation refers to a denial of participation in all or part of the communication.

10.18 Impersonation attacks

Impersonation attacks are launched by using other node's identity, such as MAC or IP address. Impersonation attacks sometimes are the first step for most attacks, and are used to launch further, more sophisticated attacks.

XI. CONCLUSION

Wireless Sensor Network is the major field in recent trends. WSN which collects information by sensing and though its major issues are discussed. And various types of attacks are also discussed.Major attacks in network layer are Wormhole attack, Denial of Service. The attacks which disrupts the routing, communication facilities and the network's funtioning The eligible sensor nodes are chosen depending on their power levels and association with number of nodes in transmission area. The various types of researches are also discussed in wireless sensor netwoks.

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Coal Mine Detection using Embedded System

R.Aswini, Jyothi.K.G, Neethu Johny and Prof.R.Gayathri

***Abstract:*-Using embedded system this paper monitors the presence of human beings in the coal mines. The method utilizes PIR Sensors, Microcontroller, MIWI technology, Zigbee to realize the operational parameter and intelligent monitored management of entire mining area. Embedded systems are controlled by one or more main processing cores that are microcontrollers. The key characteristic, however, is being dedicated to handle a particular task, which may require very powerful processing systems. If the presence of any human being is detected, then the micro controller transfers the signals to the PC (control room).**

***Keywords:* Micro controller, PIR sensors, MIWI, Human Detection.**

I .INTRODUCTION

In coal mine, security system is needed for human beings and for improving production because a coal enterprise is a high risk profession and technique. Here there is an unique example of mobile robots with embedded systems, from introductory to intermediate level. It is structured in three parts, dealing with embedded systems (hardware and software design, actuators, sensors, PID control, multitasking, mobile robot design (driving, balancing, walking and flying robots) and mobile robots applications. The embedded system is dedicated to specific tasks, design engineers can optimize it to reduce the size and cost of the product and increase the reliability and performance. This is the project which has been developed to detect the human in the restricted places, leakage of any hazardous gases and temperature sensors.

This paper was developed to monitor the presence of human beings. This is done by PIR sensors which are used to sense the human beings. There is also temperature sensor as well as gas sensors which are placed in the robot to improve the efficiency.

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In summary, coal mine detection using embedded system is a significant measure to safeguard the protection in a coal mine. It plays the role of disaster prevention and reduction in mine, as well as improves the productivity. The goal of this work is to summarize the existing solutions from various disciplines, to guide the creation of new system points towards future research directions.

II. SYSTEM SUMMARIZATION

Coal mine detection using embedded system mainly monitors the parameter in coal mine like gas, temperature as well as human beings. The author [5] has also proposed about the human count, location and track. Some major units involved are

1. Hardware part of the system comprises sensors, microcontroller (AT89S52), relay, robotic model.

2. Software part of the system is MC lab and Keil C.

III. SYSTEM HARDWARE STRUCTURE

The system hardware consists of a transmitter and receiver. Transmitter system involves power supply unit, sensors (pressure, PIR, temperature, gas), UART, MIWI, driver circuits. The received signal can be observed in PC.

A. BASIC STRUCTURE OF ROBOTIC MODEL

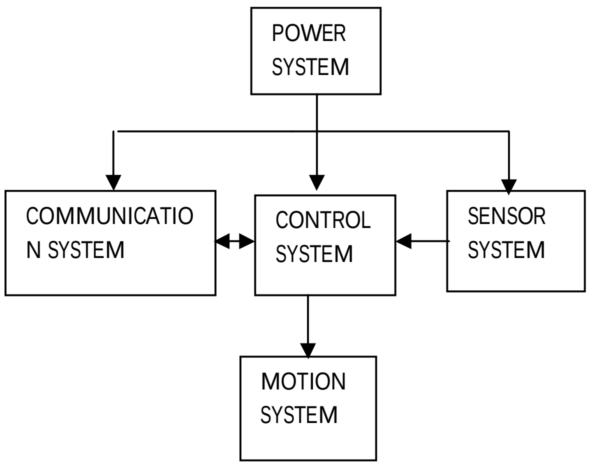

Fig.1 Basic structure of robotic model

Basically robotic model composed of power system, communication system, sensor system, control system, motion system [1].The authors have used this technology to derive the basic structure. Motion system is the trolley having all the components and a driver circuit. The communication system mainly involves the ZIGBEE which is used for wireless communication with other robots or a base station, which is concluded in [4]. Through this, wireless communication information can be easily shared between the robots, which lead to efficient context awareness for a large scale region.

B. TRANSMITTER CIRCUIT

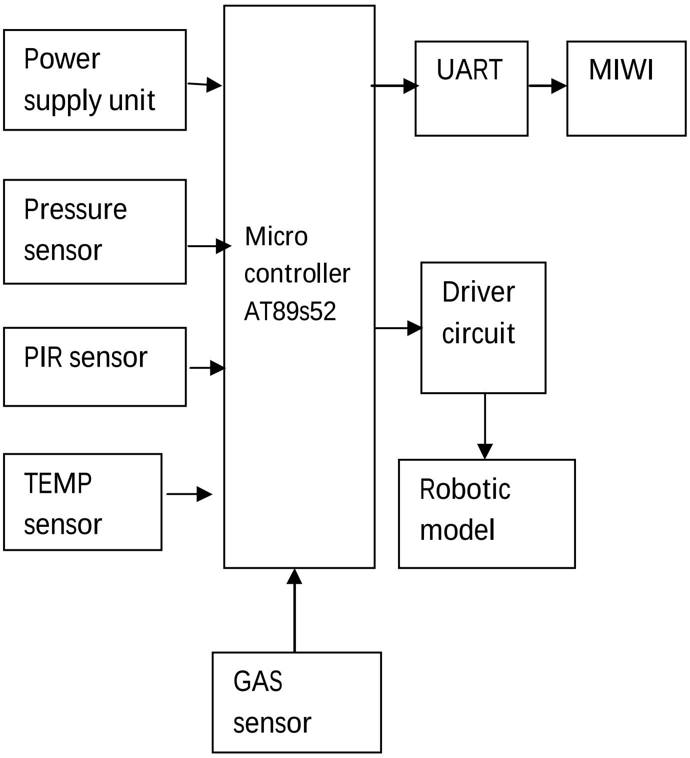

Fig.2. Transmitter

C. SENDING MESSAGES

The most important functionality of a wireless node is to communicate, or send and receive data. All protocols have reserved buffers for the data transfer, with the size equal or larger than TX_BUFFER_SIZE defined in the configuration file. Two functions are defined to manage the TX buffer in the stack: The function MiApp_FlushTx is used to reset the pointer of the transmission buffer in the stack. It has no parameter and no return value.

D. RECEIVER

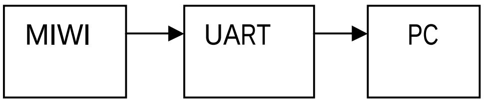

Fig.3. Receiver

E. RECEIVING MESSAGES

The other important functionality of the transceiver is to receive messages. The application layer needs to know when a message is received, the content of the message and, occasionally, how the message is received. The application layer also needs to discard the message so resources can be released and new messages can be received and processed.

IV. HARDWARE MODULE

A. ABOUT EMBEDDED TECHNOLOGY

Each day, our lives become more dependent on 'embedded system', digital information technology that is embedded in our environment. It includes not only safety-critical applications such as automotive devices and controls, railways, aircraft, aerospace and medical devices, but also communications, 'mobile worlds' and 'e-world', the 'smart' home, clothes, factories etc. All of these have wide-ranging impacts on society, including security, privacy and modes of working and living. More than 98% of processors applied today are in embedded systems, and are no longer visible to the customer as 'computers' in the ordinary sense. A new processor has been proposed[4] for the novel methods of processing, sensors, actuator, communication and infrastructures that are 'enabling' this pervasive computing. They are in a sense ubiquitous.

Embedded systems market will soon be larger than that for general purpose computing. The desktop market is stagnating; the embedded systems market is flourishing.

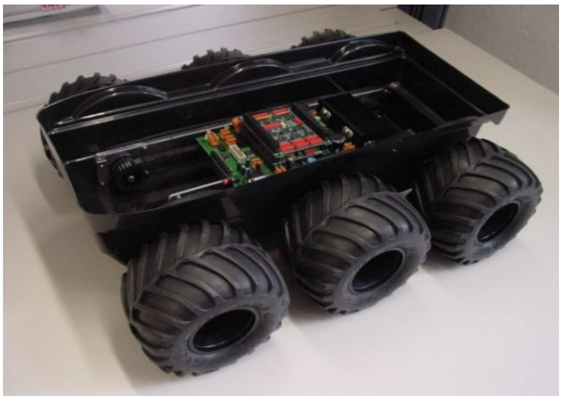

Fig.4.Trolley model

V. SOFTWARE MODULE

A. INTRODUCTION TO KEIL

The use of C language to program microcontrollers is becoming too common. And most of the time it's not easy to build an application in assembly which instead you can make easily in C. So it's important that you know C language for microcontroller which is commonly known as Embedded C.

Microcontroller Architecture support every level of software developer from the professional applications engineer to the student just learning about embedded software development. The industry-standard Keil C

Compilers, Macro Assemblers, Debuggers, Real-time Kernels, Single-board Computers, and Emulators support all 8051 derivatives and help you get your projects completed on schedule.

VI. SENSOR ALGORITHM

STEP1:

Start the program.

STEP2:

Assign the values for read, write,
Enable ports.

STEP3:

If enable=1 and set=0, condition is satisfied, first gas sensor operates for 100 seconds.

STEP4:

After 100 seconds reset occurs when enable=1 and set=1.

STEP5:

If enable=1 and set=0, condition is satisfied, at second temperature sensor operates for 100 second.

STEP6:

After 100 seconds reset occurs when enable=1 and set=1.
STEP7:

If enable=1 and set=0, condition is satisfied, at third PIR sensor operates for 100 seconds. It can sense for a distance of 5 to 6 meters.

STEP8:

After 100 seconds reset occurs when enable=1 and set=1.

STEP9:

If enable=1 and set=0, condition is satisfied, at last pressure sensor operates for 100 seconds.

STEP10:

After 100 seconds reset occurs when enable=1 and set=1.

STEP11:

In while loop four statements will be called i.e., initial, abnormal and normal.

STEP12:

If any one of the four sensors reaches the abnormal value, information will be sent to PC display.

STEP13:
The sensors again start their operation after a delay of 100 seconds by reset simultaneously after one another.

STEP14:

Stop the program.

VII. DISCUSSION

EXISTING SYSTEM:

TITLE	PROPOSED	METHOD	ADVANTAGE	OUTPUT
An intelligent coal mine security system using multi-view method	A novel monitors method for the middle and small scale coal mine.	Multiview method.	1.Real time data warning	It can distinguish security, warning and the dangerous situation with the form of sound animation, colour change and messages suggest managers to take timely measures.

Design of advanced computer controlled GPS assisted integrated unmanned robotic ground vehicle for defence operation	Self navigating system with robot to perform tasks where men cannot carry out.	Surveillance system with robotic plat form with self navigating method.	The paper delivers an entirely novel technique for nullifying the terror activities inside the building.	The designed robot vehicle has the potentialities to uproot terrorism instead of facing it and it is dedicated to save brave hearts to nation.
A rescue robot control architecture ensuring safe semi autonomous operation	The safe operation of robots under semi autonomous control of software architecture.	Standard internet/intra net technologies and cube system.	A good trade off between completely remotely operated devices and full autonomy.	All control and services are used to compute video compression.
Multi robot mapping strategy and localization	A simultaneous method of explorations which uses tactile sensors and inter robot distance measurement.	Embedded system with simulation.	1. The exploration technique expounded its independence of GPS, odometry and obstacle recognition.	1.Multitasking 2. Robots distribute themselves among obstacles in a complex environment.

PROPOSED SYSTEM:

Coal mine detection using Embedded system	**Using PIR sensors and MIWI wireless technology for sensing the presence of human in a coal mine**	**Embedded system**	**1. Affordable cost.** **2. Easy to manufacture and maintain.**	**System is implemented in coal mines for rescue purpose.**

VIII. CONCLUSION

For wireless application developers who are looking for a short range, low data rate solution, the requirements differ from point to point communication to routing messages across several hops.

The MiApp specification from microchip provides a low-cost and low-complexity solution to address nearly all those applications. It enables the wireless application developer to use Microchip's proprietary wireless protocols with little or no modification in the migration path. Working with MiMAC at the lower layer indirectly enables developers to choose any existing and future RF transceivers supported by Microchip. It is highly recommended that the readers of this application note also read application note "Microchip Wireless (MiWi™) Media Access Controller – MiMAC" (AN1283) to understand the total solution available for wireless applications from Microchip. Standardization of the lower MAC layer as MiMAC and the higher application layer as MiApp offers wireless application developers maximum flexibility in the software development process.

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Comparative Study of AI Based Gesture Recognition

Preeti.S.Ratnaparkhi and Devesh D. Nawgaje

Abstract: **Gesture recognition plays a very vital role in Human Computer Interaction. *In this paper a survey of recent hand gesture recognition systems is presented.* This research paper gives the overview of different methods for gesture recognition. *A comparative study included in this paper with focusing on different AI technique, research advantages and drawbacks are provided as well.* This research paper gives the overview of AI for gesture recognition. It also describes the process of gesture recognition using AI.**

Keywords: ***Gesture Recognition, Artificial Intelligence, Human Computer Interaction, Classification.***

I. INTRODUCTION

With the development of information technology in our society, one can expect that computer systems to a larger extent will be embedded into our daily life. People use the computer either at their work or in their spare time. The communicating with computers at this moment are limited to mouse, keyboard, track ball, web-cam, light pen and etc. But still Interaction with computers are not comfortable experience. People should communicate with computer with body language. Hand gesture recognition becomes important Interactive human-machine interface and virtual environment. A primary goal of gesture recognition research is to create a system which can identify specific human gestures and use them to convey information or for device control. Interaction between humans comes from different sensory modes like gesture, speech, facial and body expressions [2]. Gesture recognition is the process by which gestures made by the user are made known to the system. [13] Gesture recognition is also important for developing alternative human-computer interaction modalities [14]. It enables human to interface with machine in a more natural way.

Human gesture typically constitutes the space of motion expressed by the body, face or hand. Among these, hand gesture is the most expressive & most frequently used. Gesture can also be defined as a meaningful physical movement of the fingers, hands & arm or non-verbal interaction among people. We develop *gesture recognition techniques which in turn results in developing a low cost interface device for interacting with objects in virtual environment using hand gestures.*

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II. GESTURE RECOGNITION

Gesture recognition is a topic in computer science and language technology with the goal of interpreting human gestures via mathematical algorithms. Gestures can originate from any bodily motion or state but commonly originate from the face or hand.

A. TYPES OF GESTURE RECOGNITION

Gesture recognition is mainly divided into following 3 types

- Face Recognition
- Facial expression Recognition
- Hand gesture recognition

These 3 types again divided into 4 subtypes

1)Static Gesture Recognition - Static gesture have less computational complexity.

2) Dynamic Gesture Recognition - Dynamic gesture have more computational complexity.

3) On-line Gesture Recognition- Direct manipulation gestures.

4) Off-line gesture Recognition- Those gestures that are processed after the user interaction with the object.

III. Block Diagram of Gesture Recognition System

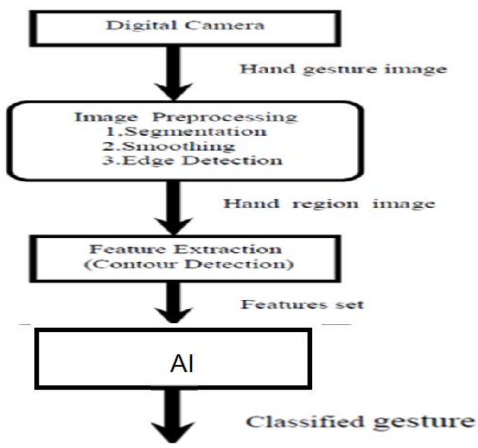

FIGURE 1 Basic block diagram of Gesture Recognition system

Collect the input: The starting point of the project was the creation of a database with all the images that would be used for training and testing. . Photographs were used, as they are the most realistic approach. Capture the images using digital camera.

Image Preprocessing: Next most important step is image preprocessing. Basically preprocessing means to extract the meaning from raw data. When the Gesture is determined,

this raw image will have to be preprocessed before it can be fed into the trained neural net for classification. One of the major limitations of neural nets is that they require a fixed number of inputs. Preprocessing must ensure that this condition is met. This input gestures are used for both training and testing purposes this phase is divided down into the following sub phases.
Segmentation
Smoothing
Edge Detection

Segmentation: The main objective of image segmentation is to extract various feature of the image which can be merged or split in order to build objects of interest on which analysis & interpretation can be performed. Segmentation is done to convert gray scale image into binary image, so that we can have only two objects in image one is hand and other is background. Segmentation is done to segment the hand area & isolate it from the background. Basically there are two methods for segmentation. These are HSV model based technique and Thresholding technique.HSV model based technique deals with the color pigment of the human skin. Thresholding technique depends on threshold value of probability. If the probability of a pixel is greater than or equal to threshold value, represents skin color. If Condition not satisfied does not represent skin color. Skin color pixels represents white and the other ones represents black

Edge Detection: Edge detection is the process of finding the meaningful transition in an image. The purpose of edge detection is to identify areas of an image where large change in intensity occurs. Intensity jump between the connected pixel is called edge and non-edge areas do not has any intensity difference. Edge detection preserves the important information in the images that represents the edges. Edge detection is usually done with local linear gradient operator .We use the Canny edge detector By finding the edge in any image we are just reducing some amount of data but we are preserving the shape.

Feature Extraction: After the preprocessing of the image and segmenting of the hand gesture, a black-white image is created and represented the hand pose inset; the feature extraction phase will start. Feature Extraction seeks to identify inherent characteristics, features of objects. These characteristics are used to describe the object, or attribute of the object, prior to the subsequent task of classification. Feature extraction operates on two-dimensional image *arrays* but produces a *list* of descriptions, or a ' feature vector'. For posture recognition, (static hand gestures) features such as fingertips, finger directions and hand's contours can be extracted. But such features are not always available due to self-occlusion and lighting conditions. Feature extraction is a complex problem, and often the whole image or transformed image is taken as input. Features are thus selected implicitly and automatically by the recognizer [9]. In this paper we select the hand contour as a good feature to describe the hand gesture shape. Contour detection process consists of two steps: first find the edge response at all points in an image using gradient computation and in the second step modulate the edge response at a point by the response in its surround [10] and extract intersection point.
second step modulate the edge response at a point by the response in its surround [10] and extract intersection point.

Artificial Intelligence: Artificial intelligence is used for classification purpose. Under AI we can use Neural Network, Fuzzy logic and Genetic algorithm, Neuro Fuzzy. But as there are many types of neural are available so different technique can be possible by using different neural network. Fuzzy rule-based methods are also used for gesture recognition. Under fuzzy mostly fuzzy clustering algorithm is used [3]. Fuzzy C-Means Clustering Algorithm shows good performance with complex background using fuzzy [3].

IV.EXISTING TECHNIQUE FOR GESTURE RECOGNITION

- **Data gloves based system**
- **Vision based system**

A. DATA GLOVES BASED SYSTEM

Data-Gloves approaches are based on the use of sensors devices, which digitize the human hand and finger movements in input parameters for a virtual reality simulation system. These methods employs mechanical or optical sensors Attached to a glove that transforms finger flexions into electrical signals to determine the hand posture [6]. Using this method the data is collected by one or more data- glove instruments which have different measures for the joint angles of the hand and degree of freedom (DOF) that contain data position and orientation of the hand used for tracking the hand [7]. However, this method requires the glove must be worn and a wearisome device with a load of cables connected to the computer, which will hampers the naturalness of user-computer interaction [5].

FIGURE 2 Data glove based system

B. VISION BASED SYSTEM:

There are few technologies already using vision based analysis system. Example: Arabic Sign Language Recognition[13]. Vision based approaches use image capturing devices. In this way a more natural interaction is achieved. a vision based gesture recognition using a simple system connected with a web camera. When the camera capture the image of hand gesture, the system extract the human hand region which is the region of interest (ROI) using the intensity color information. The system is obtained the motion velocity and the direction by tracking the center of gravity (COG) of the hand region, which provides the speed of any conducting time pattern.[8]. Vision based analysis, is based on the way human beings perceive information about their surroundings, yet it is probably the most difficult to implement in a satisfactory way. Several different approaches have been tested so far.

One is to build a three-dimensional model [18] of the human hand. The model is matched to images of the hand by one or more cameras, and parameters corresponding to palm orientation and joint angles are estimated. These parameters are then used to perform gesture classification.

Second one to capture the image using a camera then extract some feature and those features are used as input in a classification algorithm for classification [19].

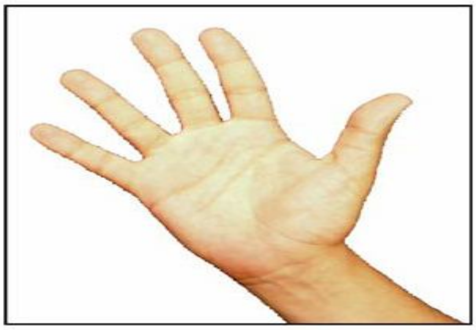

FIGURE 3 Vision based system

C. COMPARISON BETWEEN GLOVE BASED & VISION BASED METHOD

Reference	Method used	Advantage	Disadvantage	%of Accuracy
22	3D Imaging data glove & 3D electromagnetic sensor is used.	High precision technique .	Forces the user to carry a load of cables which are connected to the computer and hinders the ease and naturalness of the user interaction.	91.4
4	Web Camera is used. Supervised feed-forward neural net based training and back propagation algorithm for classifying hand gestures.	No specialized hardware is used. Only single camera is used	Feature extraction is necessary	94.6

D. COMPARISON BETWEEN AI TECHNIQUES.

Fuzzy set theory most widely used theory for soft computing, which deals with the design of flexible information processing systems, with applications in control systems, decision making, expert systems etc. Fuzzy can also be used in gesture recognition system. Fuzzy can be used for both data glove and vision based method. Mostly fuzzy is used for data glove approach. When we used the data glove method using fuzzy so to deal with the uncertainties in the data provided by the data glove, an approach based on interval fuzzy logic is used [20]. Under Fuzzy basically Fuzzy C Means algorithm is used to classify the hand gesture since it has good speed in recognizing gestures with sufficient accuracy for real-time operation[3]. To transform an expressed gesture into a meaningful statement can be a computationally intensive task and may not be easy to achieve in real-time Fuzzy Logic *[21]* can be a solution. Fuzzy Logic can help to find (and evaluate) strict aspects which describe a gesture's behavior good enough to identify it. Neural Network can also be used for classification purpose. There many types of neural network are available. If classification is accurate then testing gives perfect result. Feed forward neural network is very simple and gives very good result. Radial basis function neural network used for interpolation in multidimensional space. Neural Network can be used for both static and dynamic gesture recognition Artificial Neural Networks are one of the technologies that solved a broad range of problems in an easy and convenient manner. The working concept of Artificial Neural Networks (ANNs) is similar to human nervous system, hence it has synonym with the word neural networks. Artificial neuron is called perceptron. Neural networks, with their remarkable ability to derive meaning from complicated or imprecise data, can be used to extract patterns and detect trends that are too complex to be noticed by either humans or other computer techniques. All these neural network are used in different gesture recognition system. So each network has its own advantage and disadvantage. Some are listed below

- Feed forward Neural Network
- Radial Basis Function (RBF) Neural

- Kohonen Self-organizing Neural Network
- Learning Vector Quantization Neural Network
- Continuous Time Recurrent Neural Network
- Self-Growing and Self-Organized Neural Gas (SGONG) network
- Elman Recurrent Neural Network

Reference	Method used	Advantage	Disadvantage	% of accuracy
8	Continuous time Recurrent Neural Network is used. Use of signal predictors to recognize gestures. Tri-axial accelerometer is used to capture the gesture	Low computational cost. Use inexpensive accelerometer	Classification based on separability. Recognition accuracy depend on segmentation Signal Predictor are fast, simple and modular	94
11	Shape fitting technique is used. Self-Growing and Self-Organized Neural Gas (SGONG) network is used for classification. Skin color filtering is used for color segmentation .	The exact shape of the hand was obtained which led to good feature extraction	Input images include exactly one hand Gestures are made with the right hand only The arm must be Vertical The palm is facing the camera, and the image background is plain	90.45
16	Feed forward neural network & back propagation algorithm is used. Sigmoid activatation function used. Implemented application used for video gaming	Perform accurate classification. Structure is very simple Low computational complexity	Feature extraction is very lengthy procedure.	96.44
15	Back propagation Neural Network used for postures recognition and Elman Recurrent Neural Network For gesture recognition. Thresholding is used for segmentation. 13 data item for postures/ 16 data For gestures are taken.	Simple and active, and successfully can recognize a word and alphabet. Automatic sampling, and augmented filtering data improved the system performance.	Required long time for Learning	90
12	Hidden Markov Model (HMM) is used. Gaussian Mixture Model (GMM) Used for skin color detection. For features extraction, the orientation between the centroid points of current frame and previous frame were determined by vector	Recognized both isolated and meaningful gestures for Arabic numbers.	Recognition limited to numbers only.	89

	quantization.			
23	Radial Basis Function neural network is used. Histogram based thresholding algorithm is used. A localized contour sequence (LCS) based feature is used here to classify the hand gestures.	This network is faster Rotation invariant gesture recognition is possible	Algorithm used is very complex	99.6
3	Fuzzy C-Means Clustering algorithm is used. Implemented application is used in mobile remote. The system compares sequential images to track an object	Good performance with complex background	Performance decreases when distance greater than 1.5m between user & camera	85.83
6	Gesture recognition based on the local brightness of each block of the gesture image. Colored segmentation is done using HSV	Efficient and intuitive interaction between the human and the computer The system successfully recognized static and dynamic gestures	The database samples, variation in scale, translation and rotation led to a misclassification	91

V.CONCLUSION

In this paper various methods are discussed for gesture recognition, these methods include from Neural Network, HMM, fuzzy c-means clustering. This paper gives detailed comparison of artificial intelligence based gesture recognition system. We studied both neural network based and fuzzy logic gesture recognition system. Here we also mention advantages and disadvantages of different method. ANN provides good and powerful solution for gesture recognition. Feed forward neural network is very simple and gives very good result. Learning Vector Quantization Neural Network ration results in good classification scheme. For dynamic gestures HMM tools are perfect and have shown its efficiency especially for robot control . The single hand gesture recognition system works successfully for real-time static hand gesture recognition. From literature survey it is observed that neural network gives more accuracy as compared to the fuzzy logic approach.

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Design and Implementation of Low Voltage, High Bandwidth MOS Current Mirrors

Praween Sinha, Ajay Shankar, Mohit Arora and Mohit Datta[4]

Abstract: Design of current mirrors that can operate at higher frequency ranges is an important and growing field. This paper presents a modification to the conventional current mirrors, in the form of a precisely controlled resistance between the gates of the MOSFETs in order to achieve current mirroring at higher frequencies. Both passive and active realization techniques for the resistance have been considered in the improved current mirror. Likewise, to facilitate the operation of a current mirror at lower biasing voltages at the output, level-shifter technique has also been implemented.

I. Introduction:

In the typical IC design, biasing is often achieved by using a constant current source. Usually, this current is obtained by using a MOSFET biased to saturation, but generating separate drain currents whose values are free of process and temperature dependencies is not possible even if a known gate to source voltage is used[1]. Hence this is achieved by generating one constant and reliable current source and copying its value at different locations in the circuit. Current mirrors are employed for this purpose. Not only do they find extensive applications in almost all analog and mixed mode circuitry, their applications in analog signal processing circuits in communication systems[2] make improvement of their bandwidth an important field of study.

A Wilson current mirror is commonly used because of its good accuracy and output impedance[3]. Its operating bandwidth, although a bit higher than cascode configurations[4], can still be improved. A MOS Wilson mirror takes the circuit of Figure 2.

Applications of current mirrors in circuits operating at higher frequencies require suitable modifications because the present designs show high losses as the operating frequency is increased. In order to facilitate the use of mirrors at frequencies extending well into the microwave frequency ranges, a modification, in the form of a resistor between the gate terminals of the MOSFETs M1 and M2, has been considered. Level shifter configurations have been implemented for low voltage operation of mirrors.

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The paper is organized as follows. Section 1 deals with the simple current mirror and two modifications to it. A method for improvement of bandwidth for Wilson mirror has been presented in section 2. Section 3 presents low voltage operation circuit for a general and Wilson mirror.

1. Simple Current Mirror

A simple current mirror circuit replicates the input current of a current source and a current sink as the output current. However, the replication accuracy is poor due to channel length modulation effects. The bandwidth of this mirror is also very low.

It is a first order low-pass filter with its cut-off frequency given by[5]:

$$\omega_o = \frac{gm}{2Cgs} \qquad \text{(i)}$$

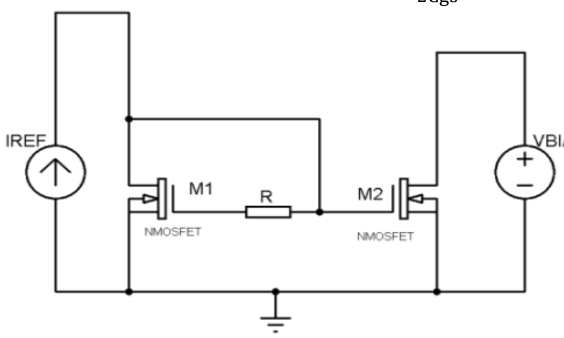

Figure 1

1. Passive Compensation: The introduction of a compensation resistor between the gates of M1 and M2 [6], as shown in Fig.1, transposes the first-order low-pass current mirror to a second-order low-pass mirror with one zero and two poles. The bandwidth is given as:

$$\omega_{o=} \sqrt{\frac{gm1}{RCgs1Cgs2}} \qquad \text{(ii)}$$

   Thus by choosing $R=1/g_m$ and $C_{gs1}=C_{gs2}=C_{gs}$, the bandwidth is given as g_m/C_{gs} which is twice the bandwidth as compared to uncompensated current mirror.

2. Active Compensation: A diode-connected enhancement type MOSFET is used as an active compensation[7] resistance which tracks the trans-conductance which varies with process and temperature drifts.

2. High Frequency Response Of Wilson Current Mirror And Small Signal Model

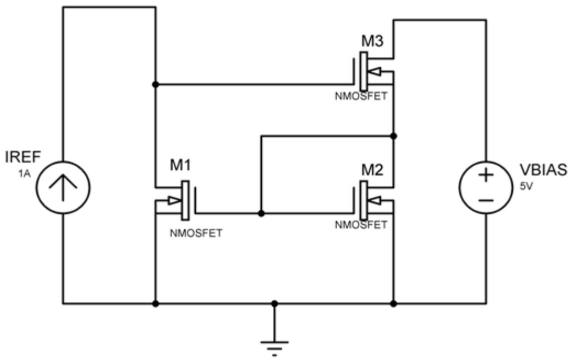

Figure 2

The transfer function of the general Wilson current mirror of Fig. 2, whose small signal model can be represented as in Fig. 3, can be obtained as:

$I_{out} = [\ g_{m2} + S(C_{gs1}+C_{gs2})\ ]\ V_{gs1}$

$V_{gs3}= (I_{ref}-g_{m1}V_{gs1}).\ \frac{1}{SCgs3}$

and, $I_{ref}= SC_{gs3}V_{gs3}+ g_{m1}V_{gs1}$

So, $\frac{Iout}{Iref} = \frac{[gm2+S(Cgs1+Cgs2)]Vgs1}{SCgs3Vgs3+gm1Vgs1}$ .... (iii)

and, $V_{gs3}= \frac{[gm2+S(Cgs1+Cgs2)]Vgs1}{gm3}$

which, upon substitution into (iii) yields,

$$\frac{Iout}{Iref} = \frac{S(Cgs1 + Cgs2) \quad + \quad gm2 * gm3}{S^2(Cgs1 + Cgs2)Cgs3 \quad + \quad SCgs3Cgs2 \quad + \quad gm1 * gm3}$$

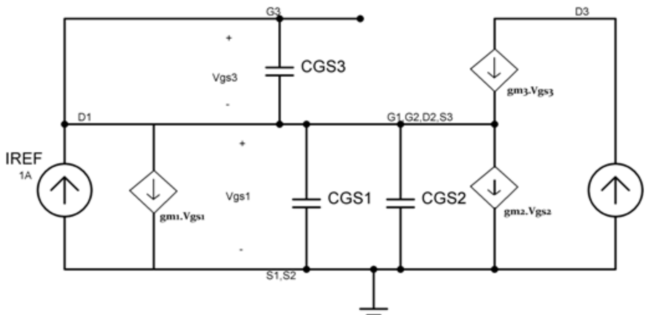

Figure 3

Hence this mirror is a second order low pass filter whose bandwidth is dependent on Cgs and the transconductance gm, and is given by:

$$\omega_o = \sqrt{\frac{gm1*gm3}{(Cgs1+Cgs2)*Cgs3}} \qquad \text{(iv)}$$

Making the assumption that process parameters are essentially equal for all transistors on a single chip, the expression for bandwidth reduces to:

$$\Omega_o = \frac{gm}{\sqrt{2}.Cgs} \qquad \text{(v)}$$

II. Enhancement of bandwidth

1. Passive compensation

Introduction of a resistance between the gate terminals of the MOSFETs M1 and M2 as shown in Fig.3 transforms the second-order low pass mirror into a third-order system with two zeros and three poles. The expression for output of the modified Wilson current mirror is given as:

$$\frac{Iout}{Iref} = \frac{S^2Cgs1Cgs2gm3R \ + \ S(Cgs1 + Cgs2)gm3 + gm2gm3}{S^3x + S^2(\ Cgs1 + Cgs2)Cgs3 \ + S(y) + \ gm1 * gm3}$$

where, x= Cgs1Cgs2Cgs3R and
Y= Cgs2gm1gm3R + gm2Cgs3

from which,

$$\omega_o^{\ 3} = \frac{gm1gm3}{Cgs1Cgs2Cgs3R}$$

$$\omega_o = \sqrt[3]{\frac{gm1*gm3}{Cgs1*Cgs2*Cgs3*R}} \qquad \text{(vi)}$$

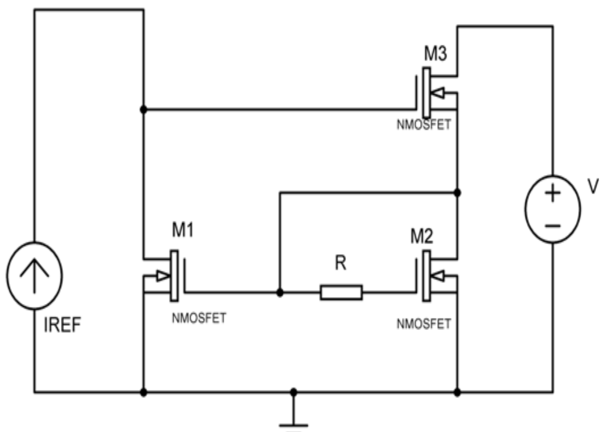

Figure 4

By selecting appropriate values of the resistance R, a considerable improvement in bandwidth can be achieved. For the value of R= $1/g_m$,

$$\omega_o = \frac{gm}{Cgs} \qquad \text{(vii)}$$

2. Active compensation

Since a passive resistor is made of poly-silicon in full monolithic integration, its value has a large tolerance. Active compensation is used so that the resistor tracks the trans-conductance of MOSFETs, which varies considerably with process and temperature drifts. This resistor, realized using the diode connected MOS transistor, will result in minimal increase in chip area and minor increase in power consumption. This also allows for significant control over the precision of the value of the resistance by varying the aspect ratio of the compensating MOSFET only. The resultant circuit is the final modified Wilson current mirror with active compensation, shown in Fig.5.

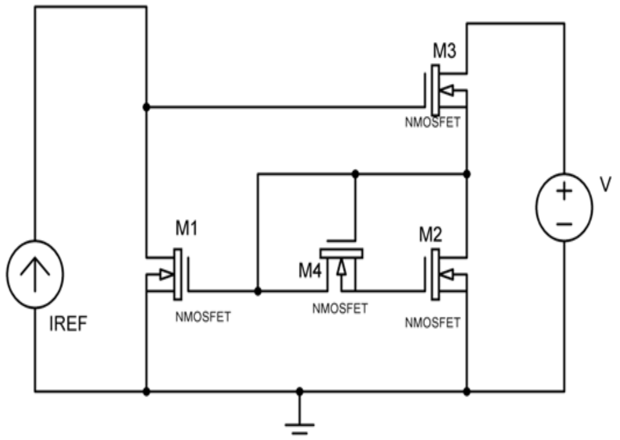

Figure 5

The previously assumed value of R= 1/gm is easily obtained here by from the MOSFET itself. Again, the expression obtained for 3dB frequency is same as before:

$$\omega_o = \frac{gm}{Cgs}$$

The value of gm can be modified to further improve the bandwidth of the mirror. This is only achieved at a compromise of chip size and area occupied by the circuit. Since using MOSFETs with channel widths beyond certain limits is unacceptable for the evolution of VLSI design, this may be achieved by using two or three MOSFETs in parallel in order to obtain higher desired values of g_m.

3. Low Voltage Analysis of the Wilson Current Mirror Using Level-shifter Technique.

Modern VLSI systems now operating from single 3.3V supplies and dropping, require high performance current mirrors that can operate with low voltages. The fact that the $V_{GS} - V_T$ of an MOS transistor determines most of its important parameters (e.g. g_m , w_o) has been used in some topologies to eliminate the dependence of the required minimal output voltage and hence increase the allowable signal swing[8]. This voltage drop (usually in the order of a volt) across the input terminal of those current mirrors may not be tolerable in all low-voltage applications. The use of current mirrors with low input voltage is especially important for implementation of VLSI test circuits which employ current sensing techniques. The level-shifter V_{GD}, can be implemented as shown in Figure. 6. The addition of a dc voltage source allows the MOSFETS to remain in saturation, hence requiring less bias voltage at the output.

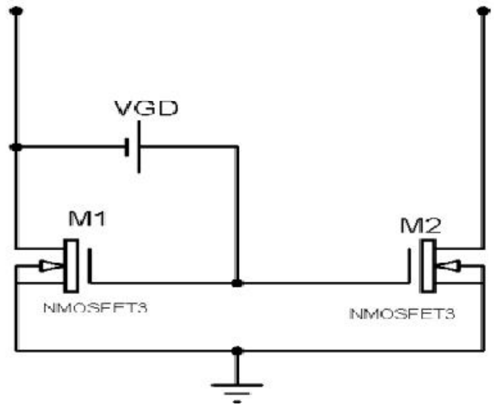

Figure 6

Suitable modifications to this circuit will need to be made, since producing a consistent voltage source V_{GD} is not a proper option. One of the means by this can be done is by using a PMOS in place of the voltage source, the gate of which is controlled by the input current itself. The analysis results have been compiled in the form of a table and simulation charts.

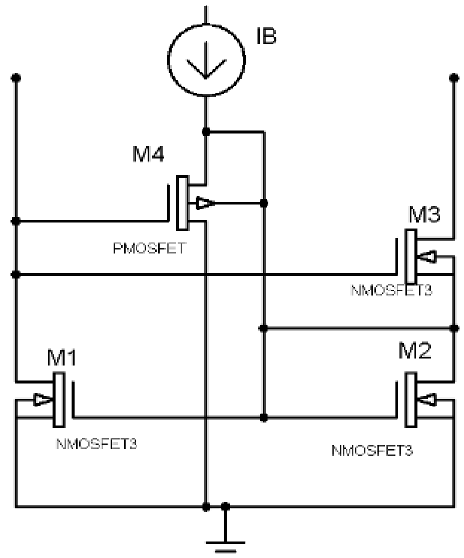

Figure 7

Level shifter technique has also been implemented for Cascode current mirrors[9], which usually require higher bias voltages to operate. However, the results obtained are much less significant than those of simple and Wilson mirrors in similar conditions.

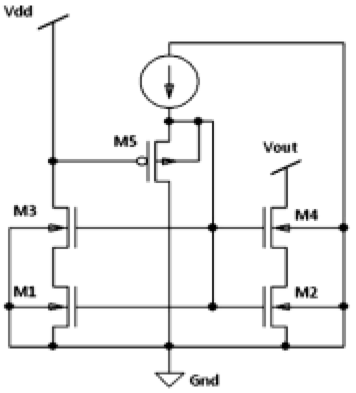

Figure 8

III. Simulation results

The results of Spice simulations are shown in charts in the following page. The Wilson current mirror in Fig. 2 has a 3dB cutoff frequency of about 762MHz, as can be seen from Chart 1. With the addition of active load compensation of this paper, the 3dB cutoff frequency for the modified Wilson mirror crosses 784MHz, i.e. an improvement of 22MHz, given in Chart 2.

Chart 3 shows improvement caused by just doubling the value of g_m in the MOSFET model. The cutoff frequency is now a little over 794MHz, resulting in a net improvement of about 32MHz. More bandwidth can yet be obtained by using still higher values of gm, as is shown in Chart 4.

The results of applying the level shifter configuration to the two versions of Wilson current mirror presented earlier are shown in charts 5 and 6. The use of level-shifting configuration facilitated precise current mirroring operation even at very low biasing voltages (2V). Thus, the voltage and power requirements of the analog IC get reduced by a great extent. It is noteworthy that the mirroring precision remains acceptable only for lower values of current, and it is poorer in the modified Wilson current mirror.

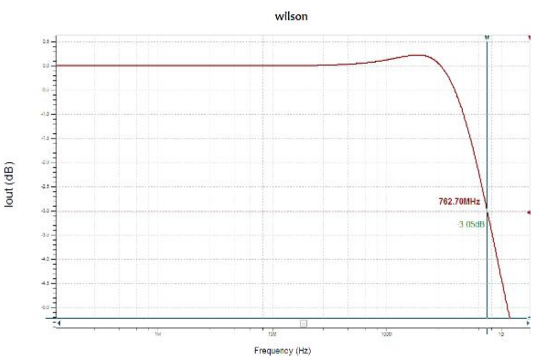

Chart 1: Frequency response of Wilson current mirror

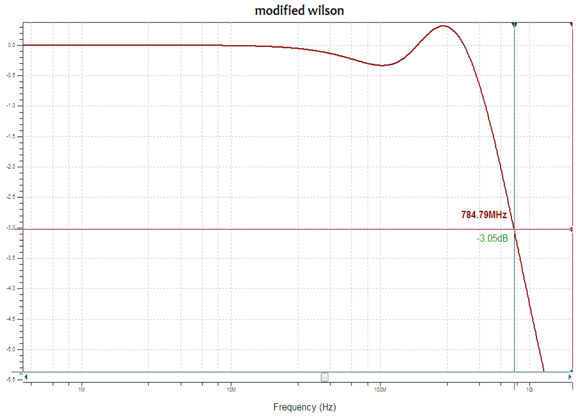

Chart 2: Response with active compensation

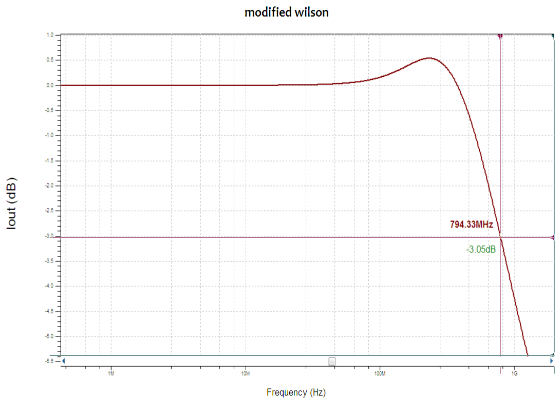

Chart 3: Response using twice the w/l ratio in active compensation

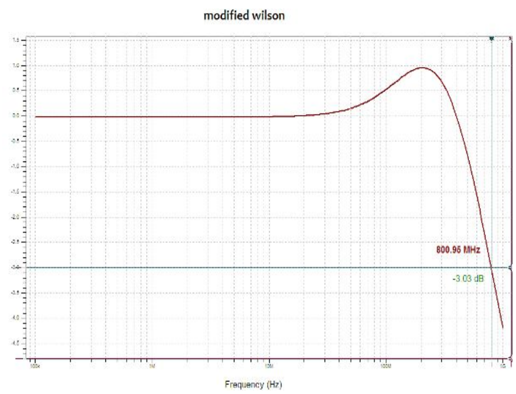

Chart 4: Response by increasing the w/l ratio by 4 times.

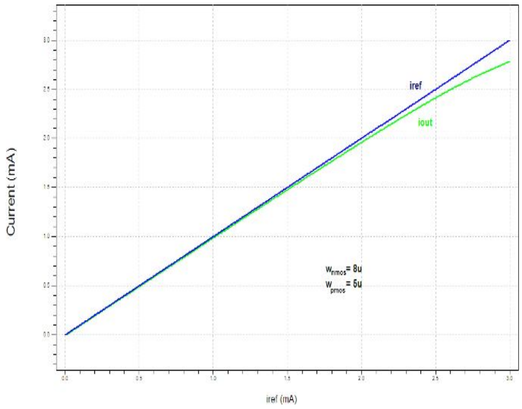

Chart 5: DC mirroring characteristics of level-shifted Wilson mirror of Fig. 7

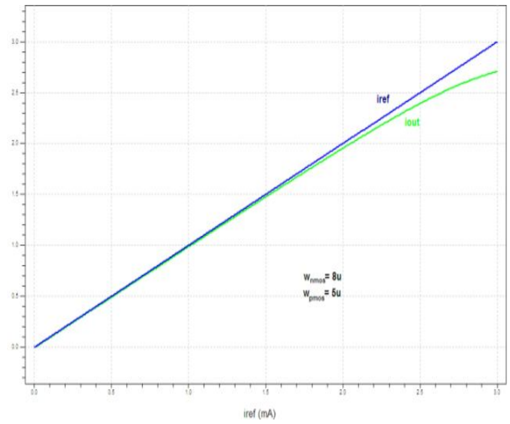

Chart 6: DC mirroring characteristics of modified Wilson mirror of Fig. 5 with level shifter applied to it

Table: Comparative results for different current mirror topologies.

Current Mirror	without resistance		with passive resistance		with active resistance	
	Bandwidth MHz	Power Dissipation mW	Bandwidth MHz	Power Dissipation mW	Bandwidth MHz	Power Dissipation mW
Simple Current Mirror	210.2	1.16	270.5	1.2	292.4	1.11
Wilson Current Mirror	762.70	4.1	784.79	4.7	794.33	4.5
Cascode Current Mirror	450	4.52	562	4.9	562	4.6
Level Shifted Wilson Mirror	762.70	1.2	784.79	1.5	794.33	1.7

IV. Conclusion

An improvement in the form of a compensation resistor between the gate terminals of two MOSFETs in the conventional Wilson current mirror has been proposed and studied. Both active and passive realization of the resistor has been considered. The active resistor is known to be easier to fabricate with a precisely controllable value, and offers potential for more bandwidth.

Likewise, low voltage operation of the current mirrors has been achieved by the use of level shifter technique in all the configurations. The biasing voltage requirement reduces to a good extent by the use of another p-MOS transistor which provides the required gate voltage to the primary transistors to achieve mirroring action. However, the use of level-shifter technique is subjected to an important limitation- the required increase in aspect ratio of the n-MOS transistors to achieve greater precision.

Taken together, these findings support strong recommendations to the use of Wilson current mirror configurations in low voltage and higher bandwidth applications and open up new possibilities which may be further studied and improved upon.

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Implementation of Graphic Simulator Based 3 Axis Robot

Miss R. Priyadharshini and Mr B. Arivu Selvam

Abstract - **Robot, which reduces the mechanical actions of the human, are widely used in industrial and military applications. There are many ways for controlling the robot using serial and wireless communication. Traditionally, much technical knowledge are required for controlling the robot. For making the controlling of robot easier, user friendly is proposed. As only roboticist has a detailed knowledge of how to communicate with robots in their language, in this paper a graphical user interface that allows easy, real-time control of robots is proposed. In this method, the graphic simulator is developed, which is considered as GUI for user, is used to command and control the robot wirelessly. GUI is operated by a mouse. By moving the mouse, the corresponding command leads to movement of robot such as go forward, go backward, left turn, and right turn which is connected through wireless communication. Graphic simulator is developed as a 3 Dimensional picture in the LABVIEW software environment.**

Keywords: **GUI (graphical user interface), virtual robot, microcontroller, receiver, transmitter, Labview, virtual instrument(VI)**

I. Introduction

A robot is a mechanical or virtual artificial agent, usually an electro-mechanical machine that is guided by a computer program or electronic circuitry. Robots have replaced humans in the assistance of performing those repetitive and dangerous tasks which humans prefer not to do, or are unable to do due to size limitations.

Robotics area has been important grown in last decades. Different kinds of robotics mechanism have emerged to perform a wide variety of task. Inside this developed, articulated robot begin to play an important role on different applications, for example: manufacture process, especially in research and teaching process. The interest on development of these robots is supported besides on electronics and computation grown. Particularly, computer graphics allow the user to build realistic environments for every problem that we wish. These changes could not be made at profitable costs and timely results without the use of computer simulation tools, since not only the robots, but also the environment must be setup several times to the desired initial conditions to repeat the experiment.

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Due to the process results, an original environment could be non-reconfigurable without human intervention (e.g. drilling a piece, moving parts to another place).One solution to the depicted issue is to account on graphical (morphological) and mathematical models to determine the interactions between robots and its external environment and then execute a desired task with intelligent (reactive or planned) obstacle avoiding in the trajectory.

A graphical user interface (GUI) that allows easy, real-time control of robots .The user uses the mouse to move the robot. i.e. Rather than typing to get the robot to move, the mouse can be used [6].

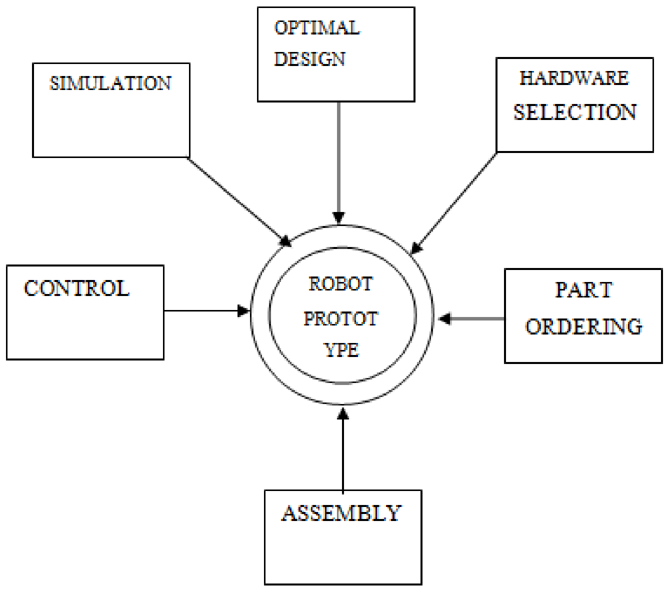

Fig. 1 Schematic view for Robot prototype

User friendly graphical interface complete with windows, menus, icons and so forth. As well as standard window technique the geometry of the model is visualized in interactive 3D viewers [3]. Using interaction with the 3D viewers the objects can be selected and manipulated. Manipulation options range from geometrical placement of the objects to adjusting the parameters of the objects.

With the help of Graphic simulator, 3D-map representation is transparent to the user, who only must concentrate the effort on its own application. Hence, the user leaves the usually graphical part which acts as a true virtual cell with robots reflecting the actions of user's application.

II. EXPERIMENTAL SETUP

SOFTWARE:

Labview ties the creation of user interfaces (called front panels) into the development cycle. Labview programs/subroutines are called virtual instruments (VIs) [8]. Each VI has three components: a block diagram, a front panel and a connector panel. The last is used to represent the VI in the block diagrams of other, calling VIs. Controls and indicators on the front panel allow an operator to input data into or extract data from a running virtual instrument. However, the front panel can also serve as a programmatic interface. Thus a virtual instrument can either be run as a program, with the front panel serving as a user interface, or, when dropped as a node onto the block diagram, the front panel defines the inputs and outputs for the given node through the connector panel.

Labview used here is to create the 3D picture of the robot which is going to be controlled [1]. When the 3D image is operated by the mouse in the transmitter side, the command is given to the micro controller through the serial communication , which in then rotates the motor driver in the receiver side.

Robotics simulation programs in Labview include the following items:

Robotics simulator: The simulator reads and displays simulation scenes which was designed by calculates realistic, physics based properties of simulated components as they interact, an advances the time in the simulation. Start the simulator in a VI .

Simulation program: Create programs to run in the simulator. These programs contain the following components:

Manifest file: After designing a simulation environment and components, save their definitions in a .xml file, called a manifest file. The simulator reads from manifest files to render the components they define.

Simulation scene: Each manifest file defines one simulation scene, a combination of simulated components and their properties.

Environment: Each simulation instance must have an environment that describes the ground and any attached features. The environment also has associated physical properties, such as the surface material and force of gravity.

Robots: Robots can contain the base and the joints

VIs: Write VIs to control the simulator and simulated components. The VIs contain the same code to control simulated robots that embedded applications running on real robots might contain.

Display window: When starts the simulator, the simulation displays in the 3D picture control

HARDWARE:

Basic elements of this project are as follows:
Microcontroller, Encoder, RF transmitter, Decoder, RF receiver [5].

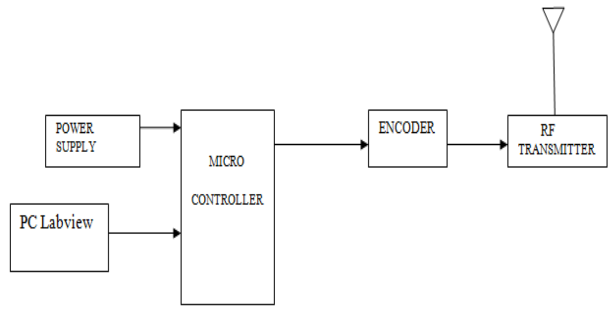

Fig .2(a):Transmitter section

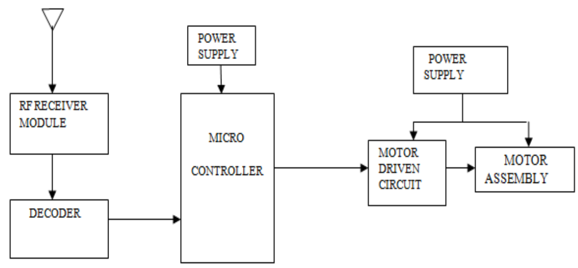

Fig. 2(b): Receiver section

Transmitting Section: After proper signal conditioning the microcontroller gives its output to port no.P0. Then these output signals behave as a feeder to relays . This is driver for operational four relays. This IC act as a source to drive the relay parts and finally the signal is transmitted through, transmitting IC. This IC provides proper modulation to our control signal to be transmitted through max proper; here the crystal oscillator is used as a source of carrier generator

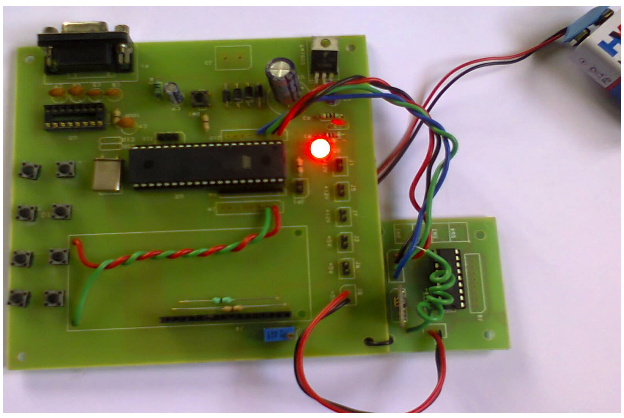

Fig 3: Modulation section

Receiving Section: Receiver IC receives the control signal and this IC provides the demodulation of carrier signal. Then again the output of receiver is amplified and filter up to the proper dc level. The signal acts as a source for the DC motor to provide the control movement. Then the whole process governs in this way.

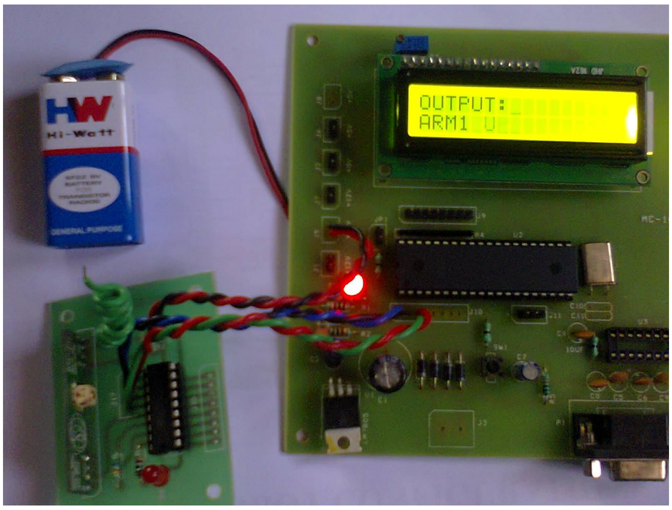

Fig 4:Demodulation section

The main aim of the project is to provide the data transmission from the GUI and microcontroller through serial communication and to provide the data transmission between the GUI and Robot wirelessly [2].

The main task of this project is two parts: (1) to program the microcontroller on both the transmitter section and the Robot interfaced to the radio packet controller module which would enable us to wirelessly control the Robot; (2) to program the GUI Application which would enable us to serially control the transmitter section.

III. Conclusion and Future Work

The graphic simulator was developed by creating the 3D picture control in Labview environment. This graphic simulator is able to control the 3 axis robot wirelessly by the user control.

The group of robots will be developed as 3D image and controlled wirelessly.

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Improving Robustness of Video Watermarking Scheme Using DWT Domain

Ms. Snehal Pritesh Shah

***Abstract*- In this paper, a algorithm based video watermarking scheme in the discrete wavelet transform (DWT) domain is proposed. Scene change analysis is first conducted to decompose video into different scenes. Each frame of the video is transformed to wavelet domain by DWT. The watermark image is decomposed into 8-bit planes, scrambled and embedded into the mid-frequency DWT coefficients. The quality of the watermarked video is enhanced by GA. Experimental results demonstrate that it is robust to common attacks in video watermarking such as frame dropping, frame averaging additive noise and lossy compression.**

***Keywords*— Abrupt scene change, peak signal noise ratio (PSNR), video watermarking, discrete wavelet transform, robustness.**

I. INTRODUCTION

Digital watermarking is an effective way to protect copyright of multimedia data even after its transmission. Watermarking is a concept of embedding a special pattern, watermark, into a multimedia document so that a given piece of copyright information is permanently tied to the data. This information can later prove the ownership, identify a misappropriating person, trace the marked document's dissemination through the network, or simply inform users about the rights-holder or the permitted use of the data. Most of the proposed video watermarking schemes are based on the techniques of image watermarking and directly applied to raw video or compressed video..Video watermarking introduces some issues which is not present in image watermarking. Due to a large amount of data and inherent redundancies between frames, video signals are highly susceptible to attacks, such as frame averaging, frame dropping, lossy compression and statistical analysis. However, the currently available algorithms do not solve these problems effectively. Since the goals of watermarking, such as robustness, transparency and capacity, are usually conflicting, GA can be utilized to solve the optimization problem. video watermarking schemes must not use the original video during watermark detection as the video usually is in very large size and it is inconvenient to store it twice. We propose a new watermarking scheme to overcome these problems. The watermarking process overview is depicted in Fig. 1. Video sequence is pre-processed by scene change detection and DWT.

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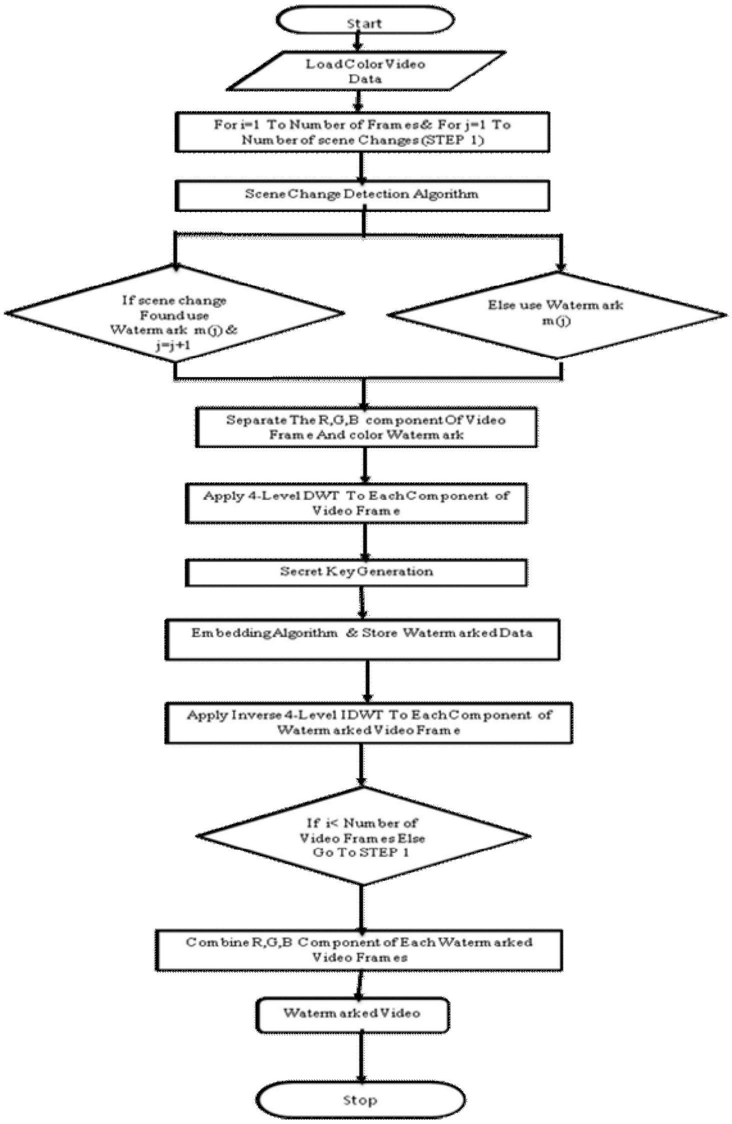

Fig.1 Overview of watermarking process

First of all original color video is stored in one array then it is converted in to frames. Then scene change detection algorithm is applied on to the consecutive frames. If scene change found we have to change the watermark image else apply the same watermark within one scene of the video. Decompose video frame and watermark in to three different components. apply the 4 level DWT on each component of video frame. generate the secret key and add in to watermark data because watermark image bits are not enough for embedding in to original image. Apply the embedding algorithm for watermark embedding and store the watermarked data in to one array. Apply the 4 level inverse DWT to watermarked video frames. Repeat all the step up to number of frames. Combine R,G,B component of each watermarked video frames. so we get the watermarked video.

II. WATERMARK IMAGE PRE-PROCESSING

A watermark image is pre-processed and embedded into different scenes, so that it can resist a number of attacks toward the video. It is firstly decomposed into binary image. The watermark binary color image is decomposed in to three components (Red, Green and Blue) and these three components are applied to the secret key generation algorithm. Finally three watermark images are generated which will be embedded in three different components of original color video frame.

length of generated watermark image = length of original watermark

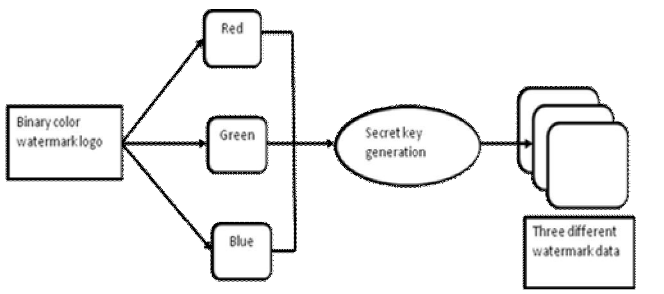

Fig.2 Overview of watermark preprocess

To further increase the security of the watermark, it is scrambled using Arnold transform, which is in fact an iterative process to move the pixel position . It is defined as follows:

$$\begin{Bmatrix} X' \\ Y' \end{Bmatrix} = \begin{Bmatrix} 1 & 1 \\ 1 & 2 \end{Bmatrix} \begin{matrix} X \\ y \end{matrix} \mod N$$

where (x, y) is the pixel position, and (x', y') are the new position after Arnold transform. N is the pixel width or height.

III. VIDEO PRE-PROCESSING

A. Scene change detection

Since applying a fixed image watermark to each frame in the video is hard to maintain statistical and perceptual invisibility and applying independent watermarks to each frame cannot resist statistical comparison or average. The new watermarking scheme is based on scene change detection, which is to decompose video sequence into different video scenes. Independent watermarks are employed for successive but different scenes and an identical watermark is embedded within each motionless scene. Histogram analysis is utilized for scene change detection. If the histogram difference of two scenes is greater than the threshold, we consider there is a scene change.

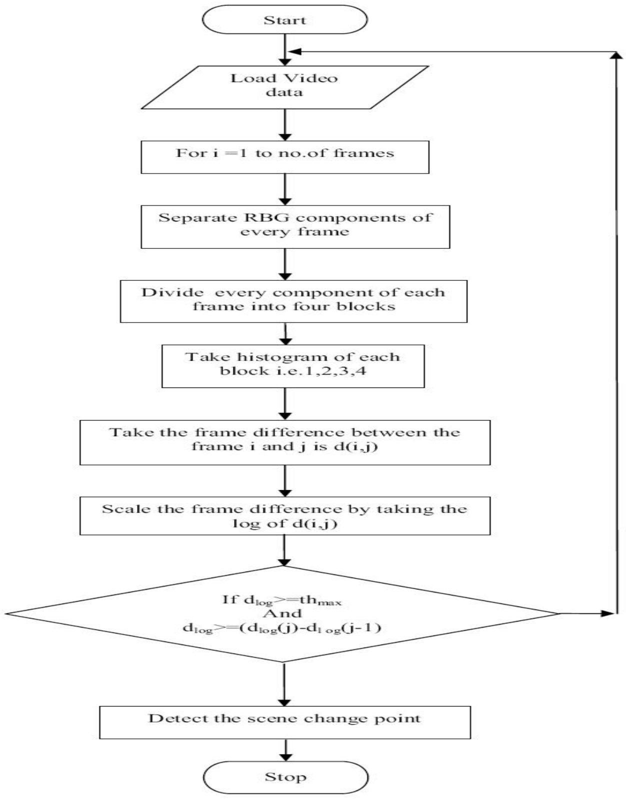

Fig.3 Flow chart for abrupt change in video transition

B. DWT

The luminance components of the input video frames are transformed to frequency domain and the middle-frequency range coefficients of the watermark are modified according to the watermark. The basic idea is that the human eyes are sensitive to the low frequency noise and the quantization step of lossy compression may discard the high frequency components. Therefore, the reasonable trade-off is to embed the watermark into the middle-frequency range of the video frames. Each video frame is transformed to the wavelet domain with 4 levels. Haar wavelet is used for simplicity, and only the LH1, HL1, LH2 and HL2, LH3, HL3, LH4, HL4 coefficients are embedded with scrambled watermark. Figure 4 shows the result of the “Lena” image after a four-level 2-D DWT.

Fig.4 Lena image for Testing

Fig.5 "Lena" image after a four-level 2-D DWT

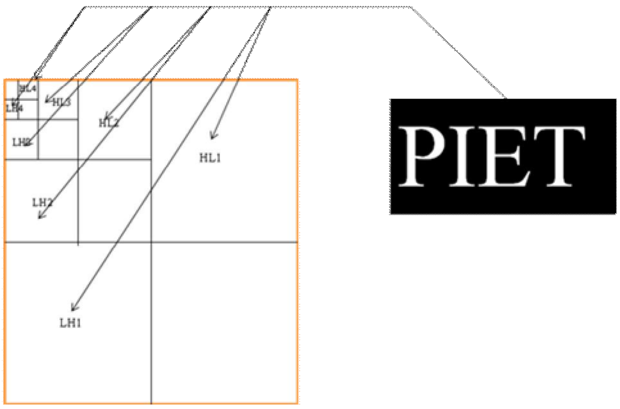

Fig-6 embedding process

In addition, with such a scheme, it is not possible to add more watermark energy at a particular frequency, in which the image energy is high, in order to improve robustness. LL band contains the most of energy of the image ,so we apply the watermark in mid frequency bands ,it will not creates some artifacts in the image. and invisibility of the watermark can be increased.

IV. WATERMARK EMBEDDING AND DETECTION

A .Watermark embedding

Figure 7 Shows how to embed the Watermark and secret key in to the video frames here we apply the watermark in to only mid frequency band of the image because of the effect of the human visual system. The power present in the frequency bands varies greatly from image to image. If the image energy in a particular band is very low and the watermark energy in that band is high, then some artifacts are created in the image, since the watermark energy is too strong relative to the image.

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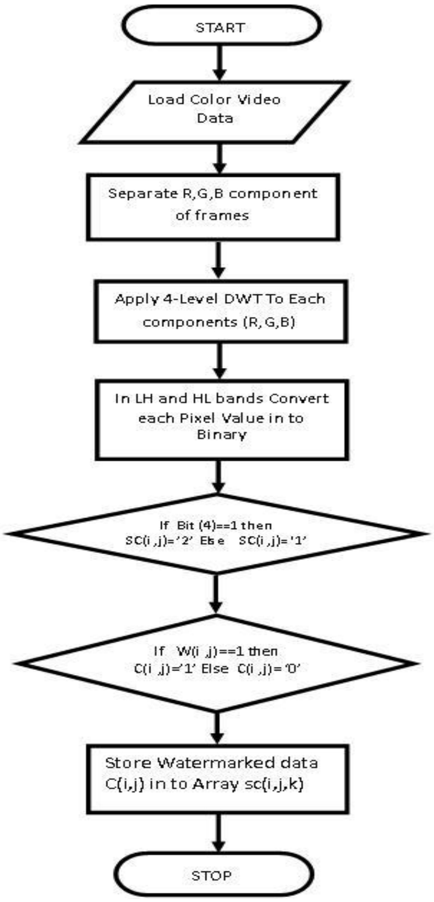

Fig. 7 Flow chart for watermark embedding

Figure 7 shows the Flow chart of embedding algorithm to embed the watermark in to color video. first of all convert the video in to frames and then frame to image. decompose each image in to three color components (R,G,B). Apply 4-level DWT to each component. Here we apply watermark in to HL and LH bands (mid frequency bands) of each level so convert each pixel value in to binary. To increase the robustness we apply the watermark in the 5th bit of every pixel. If 5th bit is equal to 1 then SC(i , j ,k) is equal to 2 else SC(i , j ,k) is equal to 1, where SC is the array to store the information for original video Extraction and (i,j) is the pixel position of particular band and k is the frame number. Then if W(i , j) =1 then 5th bit of pixel is equal to 1 else it is 0.we start to embed the watermark from HL4 (4th level mid frequency band) and then sequencely in to LH4, HL3, LH3, HL2, LH2, HL1 and LH1. At last store the watermarked video frame data in to array sc(i, j, k) for watermark Extraction before applying inverse DWT.

C. Watermark detection

The extraction process requires the key used for selecting the frames, the wavelet transform filter, and the channel I which the watermark is inserted. Due to the act of requantizing, a threshold region T should be defined in order to detect the existence of the video watermarking.

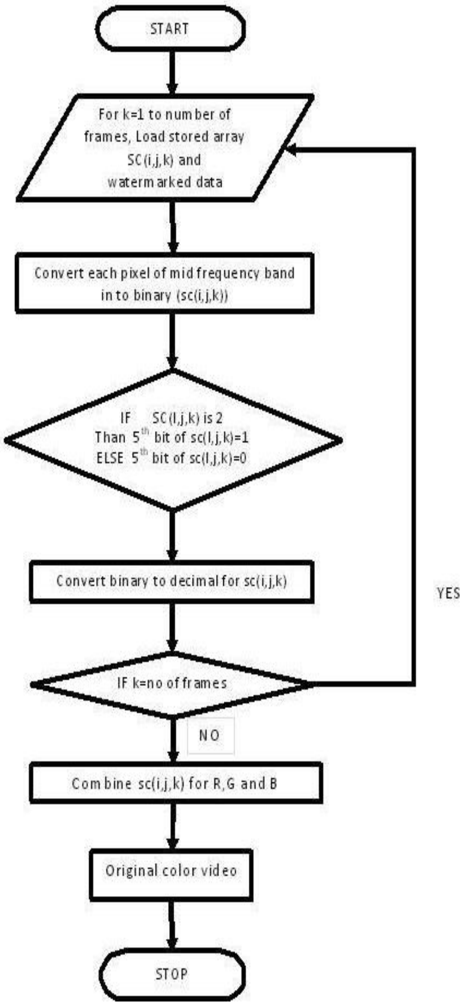

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Fig.8 Flow chart for watermark and secret key Extraction

Figure 8 shows the Flow chart of extraction algorithm to extract the watermark and secret key from the watermarked video. First Load the watermarked data stored in array sc(i,j,k) ,where (i,j) is the pixel position in particular frequency band and k is the frame number. Let convert each pixel of mid frequency band in to binary. Check if 5th bit of pixel value is „1" then WR (i,j)==1 ,where WR is the recovered watermark and secret key matrix. if 5th bit of pixel value is „0" then WR(i,j)==0. Decompose watermark data and secret key from the matrix WR. Apply these algorithm to three different watermarked images so we can get three different watermark data, then combine these data to retrieve the watermark image

V. SIMULATION RESULTS AND DISSUCTION

Fig 9- original video frames

Fig 10-watermarked video frames

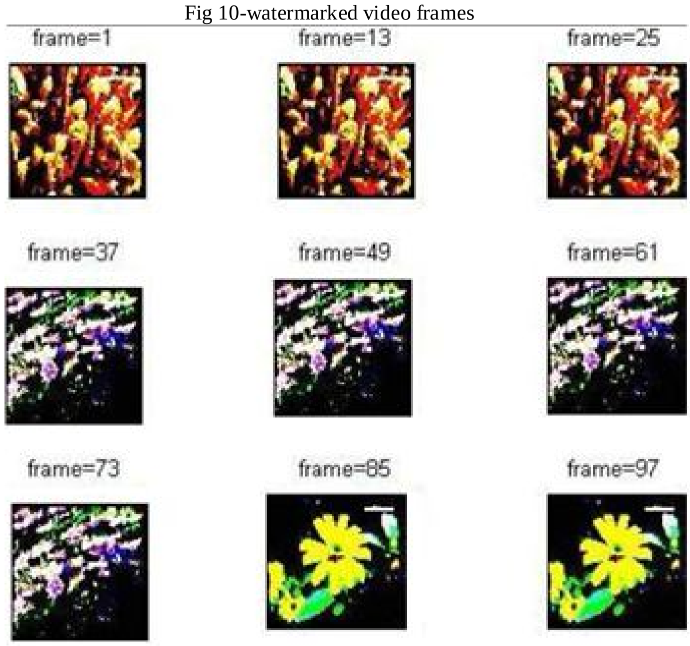

Fig 11. Recovered watermark image

In order to evaluate the quality of image, we use parameter peak value signal-to-noise ratio (*PSNR*)

$$\text{PSNR in db} = 10\log_{10}\frac{255^2}{\frac{1}{N\times N}\sum_{i=1}^{N}\sum_{j=1}^{N}[f(i,j)-f'(i,j)]^2}$$

where N is the size of image, $f(i, j)$, $f'(i, j)$ is the pixel gray value of host image and pending detection image respectively. The bigger the value of *PSNR*, the better the quality of image.

In order to evaluate the robustness of watermarking algorithm, the comparability between original watermark w and detected watermark w^* is calculated with the formula given in Eq

$$NC = \frac{\sum_{i=1}^{N}\sum_{j=1}^{N} w(i,j)\times w^{*},(i,j)}{\sum_{i=1}^{N}\sum_{j=1}^{N} w(i,j)^2}$$

Here the *NC* value is 1 for all images, This shows the 100% recovery of watermark images , the *PSNR* value is between 33 to 41.50 *db*. The *PSNR* value shows that the algorithm keeps the quality of the image and invisibility of embedded watermark without any attacks.

VI. CONCLUSION

In this paper I propose a scene-based watermarking scheme. The scheme is robust against various attacks because we does not require original video as well as watermarked video for original video and watermark video recovery. Experiment has been done on these novel video watermarking schemes to test an show its performance. The robustness of our approach is demonstrated using the calculation of NC.

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Knowledge Engineering – Knowledge Management, an Interface

Ms. Tiny Tanushree Gohain and Dr. R. Jayam

Abstract: **The disciplines of knowledge engineering and knowledge management are closely tied. Knowledge engineering deals with the development of information systems in which knowledge and reasoning play pivotal roles. Knowledge management, a newly developed field at the intersection of computer science and management, deals with knowledge as a key resource in modern organizations. Managing knowledge within an organization is inconceivable without the use of advanced information systems; the design and implementation of such systems pose great organization as well as technical challenges. This article will first give an overview of some important historical developments in KE: special emphasis will be put on the paradigm shift from the so called transfer approach to the so called modeling approach. This paradigm shift is sometimes also considered as the transfer from first-generation expert systems to second-generation expert systems. This study will also present some modeling frameworks which have been developed in recent years such as CommonKADS, MIKE and PROTE'GE'II. Problem solving methods have been a major research topic in KE for the last decade and so, the characteristics of problem-solving methods are described. Ontologies, which gained a lot of importance during recent years are also discussed. The paper concludes with a discussion of current developments in KE and its relationship to Knowledge management.**

Keywords**: Knowledge engineering, Knowledge management, Modeling approach, Ontology, Transfer approach,**

I. INTRODUCTION

In earlier days research in Artificial Intelligence (AI) was focused on the development of formalisms, inference mechanisms and tools to operationalize Knowledge-based Systems (KBSs). The development efforts were restricted to the realization of small KBSs in order to study the feasibility of the different approaches. Knowledge Engineering (KE) is similar to that of Software Engineering: turning the process of constructing KBSs from an art into an engineering discipline. This requires the analysis of the building and maintenance process itself and the development of appropriate methods, languages and tools specialized for developing KBSs (G. Shaw, B.R. Gaines, 1992). In this present era of technology, most current knowledge management activities rely on database and Internet systems. If knowledge is stored explicitly at all, it is typically in databases either as simple tables (for example, relational databases) or semi-structured text (as in Lotus Notes). The use of sophisticated knowledge representation systems such as Classic, Loom, or G2 is rare. Also, few organizations have a systematic process for capturing knowledge. Success in the marketplace is increasingly linked to an organization's ability to manage and leverage its intellectual capital, the intangible and often invisible assets such as knowledge and competence of people, intellectual property, and information systems that don't show up directly on the bottom line but are at least as valuable as financial assets. Successful companies of the 21st century will be those who do the best jobs of capturing, storing, and leveraging what their employees know -Lewis Platt, CEO, Hewlett Packard.

II. OBJECTIVES OF THE STUDY

- To provide historical developments in KE emphasizing specially on the shift from transfer approach to that of modeling approach.
- To present some modeling frameworks which have been developed in recent years.
- To study on the problem solving methods.
- To discuss on the current developments of Knowledge Engineering.
- To analyze the relationship between Knowledge engineering and Knowledge Management.

III. LITERATURE REVIEW

Recent studies have adopted a variety of perspectives to deal with KM activities: (1) a design perspective that focuses on the creation and the evolution of knowledge G. Fischer, J. Ostwald, 2001); (2) an information technology perspective that discusses how KM and IT relate to each other (D.E. O'Leary, 1998 and L.T. Wilson, C.A. Snyder, 1999); (3) a management perspective that investigates critical factors and challenges in KM (L.L. Kemp, K.E. Nidiffer , 2001); (4) an artificial intelligence perspective that emphasizes the role of artificial intelligence in KM (J. Liebowitz, 2001); and (5) an ontology perspective that focuses on developing common formalisms to enable knowledge sharing (D. Corbett, 2004 AND D. Fensel, 2002).Traditional KM treats knowledge workers as passive recipients of information and its goal is to store information from the past so that lessons will not be forgotten. Fischer and Ostwald, 2001 recognized that future information needs will be different from past needs and proposed a design perspective approach for KM. In this

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framework, KM is a cyclic process involving three related activities: creation, integration, and dissemination.

The design perspective shares three essential viewpoints: (1) KM relates to working, learning, and knowledge creation; (2) workers, not managers, create knowledge at the time of use; and (3) knowledge is a side effect of work. Several barriers to implementing a design-oriented KM approach are pointed out. First, a common understanding of knowledge is required among knowledge workers for sharing and interoperating knowledge. Second, KM must support the continuous change of knowledge. Third, to address information overload, finding information relevant to the task at hand is becoming increasingly critical.

L.T. Wilsonn and C.A. Snyder (1999) considered that successful KM has three dimensions: information technology, people, and processes. Their research focuses on the IT aspect of KM. Two types of knowledge are classified: support information and guidance. The support information includes descriptive explanations that serve as a basis for understanding (who, what, when, where, with what, and why). The guidance strengthens the ability to act, through explanations of how to accomplish a task. IT plays a major role as a medium for delivering knowledge. To support KM successfully, the IT applications must embrace both support information and guidance and give users an interface to access both types of knowledge selectively and at the point of need.

O'Leary , 1998 treated KM as a set of converting and connecting processes. The classic KM approach only stresses converting employees' private knowledge to generally available public knowledge and connecting knowledge and people. He advocated that the full range of KM converting and connecting capabilities should include: (1) converting individual to group-available knowledge; (2) converting data to knowledge; (3) converting text to knowledge; (4) connecting people to knowledge; (5) connecting knowledge to knowledge; (6) connecting people to people; and (7) connecting knowledge to people. In his opinion, there were still limitations associated with focusing on knowledge. First, knowledge does not necessarily result in direct action to create value. Second, knowledge processes are continuously changing. Third, knowledge alone does not guarantee a creative response to decision-making situations.

Kemp et al., 2001 viewed KM as leveraging relevant intellectual capital to achieve organizational missions. The major goal of KM was to leverage knowledge to measurably enhance effectiveness, efficiency, and innovation. The authors uncovered a set of critical factors that constitute the essential competencies in KM. These are: (1) a systematic approach; (2) a flexible framework; (3) an evolutionary process; (4) integrated measurement; (5) a capability model; and (6) technical maturity.

Preece et al, 2001 viewed KM as the efforts to capture, store, and deploy knowledge using a combination of information technology and business processes. They believed that current KM practice significantly underutilized knowledge engineering technology. Hence, they used knowledge acquisition processes to capture structured knowledge systematically and knowledge representation technology to store knowledge. After applying knowledge engineering to implement a drilling KM system, they pointed out several problems in their system: (1) it is difficult to issue complex queries to the knowledge store; (2) it is hard to add new cases; and (3) multiuser access is limited. Liebowitz, 2001 defined KM as the process of creating value from an organization's intangible assets. KM combines concepts from numerous disciplines, including organizational behavior, human resource management, artificial intelligence, and information technology. He pointed out that KM can be advanced by applying AI techniques such as knowledge acquisition, knowledge discovery, ontologies, and intelligent agents. However, there are some potential pitfalls that could squash KM initiatives in the organization: (1) the culture of organizations; (2) the true value of KM is not realized; (3) the knowledge repositories become unwieldy and difficult to maintain; (4) security of the knowledge repositories may be compromised; (5) systems may be ill designed and difficult to use; (6) a KM program plan may be ill conceived and problematic; and (7) the employees must be motivated.

An ontology (T.R. Gruber, 1995) is a shared description of concepts and relationships in domain knowledge. It consists of terms, their definitions, axioms relating to them, and a taxonomy organizing them. The main objective of ontology is to enable communication and knowledge sharing by capturing a shared understanding of terms that can be used by humans and programs. There are many well-known representation languages for ontology definition, including Ontolingua, KIF, LOOM, RDF, DAML, OIL, OWL, DLs, FLs, and CGs [16.18,19,20,22]. Tools implemented to support the ontology development process can be grouped into six clusters: ontology development tools, ontology merge and integration tools, ontology evaluation tools, ontology-based annotation tools, ontology storage and querying tools, and ontology learning tools.

Maedche et al.,(2006) proposed an integrated KM architecture for implementing an ontology-based KM system. In the architecture, an ontology server is built based on the KAON architecture, an ontology mapping process is used to manage multiple ontologies, and an ontology evolution process is used to manage evolving ontologies. They also pointed out the important challenges to improve traditional KM systems, including scalability, managing multiple ontologies, ontology evolution, and embedding ontologies in daily work tools. To address the problems of analyzing KM processes and dynamically updating the KM solution, Staab et al. (2001) proposed an ontology-based KM methodology that consists of a knowledge process and a knowledge metaprocess.

The knowledge process focuses on handling knowledge items and includes four steps: knowledge creation and import, knowledge capture, knowledge retrieval and access, and knowledge use. The knowledge metaprocess focuses on introducing and maintaining KM systems and includes five phases: feasibility study, kickoff, refinement, evaluation, and maintenance. The On-to-Knowledge project (D. Fensel, 2002) developed an ontological methodology and tools for acquiring, maintaining, and accessing knowledge. The architecture includes OntoWrapper and OntoExtract to extract metadata from Web documents in RDF format, OIL to capture the semantics of knowledge, Sesame to offer a data repository and querying facility, and several ontology middleware modules such as an advanced reasoning engine, a knowledge-sharing facility, a search engine, a presentation platform, and an ontology editor. Corbett (2004) demonstrated that CGs provide the appropriate expressiveness for the task of representing, comparing, merging and interoperating with ontologies. The type hierarchy and the canonical formation rules specialize ontologies into concrete instances by instantiating canonical CGs from the type hierarchy. The merge operation, using join and type subsumption, can be used to perform knowledge conjunction of concepts represented in the ontologies.

IV. KNOWLEDGE ENGINEERING: DEFINITION

KE is an engineering discipline that involves integrating knowledge into computer systems in order to solve complex problems normally requiring a high level of human expertise (Edward Feigenbaum, and Pamela McCorduck,1983).

At present, it refers to the building, maintaining and development of knowledge-based systems. It has a great deal in common with software engineering, and is used in many computer science domains such as artificial intelligence, including databases, data mining, expert systems, decision support systems and geographic information systems. Knowledge engineering is also related to mathematical logic, as well as strongly involved in cognitive science and socio-cognitive engineering where the knowledge is produced by socio-cognitive aggregates (mainly humans) and is structured according to our understanding of how human reasoning and logic works.

V. VARIOUS ACTIVITIES OF KE SPECIFIC FOR THE DEVELOPMENT OF A KNOWLEDGE-BASED SYSTEM:

- Assessment of the problem
- Development of a knowledge-based system shell/structure
- Acquisition and structuring of the related *information*, *knowledge* and specific *preferences* (IPK model)
- Implementation of the structured knowledge into knowledge bases
- Testing and validation of the inserted knowledge
- Integration and maintenance of the system
- Revision and evaluation of the system.

Being still more art than engineering, KE is not as neat as the above list in practice. The phases overlap, the process might be iterative, and many challenges could appear.

VI. VIEWS OF KNOWLEDGE ENGINEERING

As per Schreiber, August Th.; Akkermans, Hans; Anjewierden, Anjo; Dehoog, Robert; Shadbolt, Nigel; Vandevelde, Walter; Wielinga, Bob (2000), there are two main views to knowledge engineering:

- Transfer View – This is the traditional view. In this view, the assumption is to apply conventional knowledge engineering techniques to transfer human knowledge into artificial intelligence systems.
- Modeling View – This is the alternative view. In this view, the knowledge engineer attempts to model the knowledge and problem solving techniques of the domain expert into the artificial intelligence system.

A major concern in knowledge engineering is the construction of ontologies. One philosophical question in this area is the debate between foundationalism and coherentism - are fundamental axioms of belief required, or merely consistency of beliefs which may have no lower-level beliefs to justify them?

Knowledge engineering is a field within artificial intelligence that develops knowledge-based systems. Such systems are computer programs that contain large amounts of knowledge, rules and reasoning mechanisms to provide solutions to real-world problems.

A major form of knowledge-based system is an expert system, one designed to emulate the reasoning processes of an expert practitioner (i.e. one having performed in a professional role for very many years). Typical examples of expert systems include diagnosis of bacterial infections, advice on mineral exploration and assessment of electronic circuit designs.

VI. IMPORTANCE OF KNOWLEDGE ACQUISITION

The early years of knowledge engineering were dogged by problems. Knowledge engineers found that acquiring enough high-quality knowledge to build a robust and useful system was a very long and expensive activity. As such, knowledge acquisition was identified as the bottleneck in building an expert system. This led to knowledge acquisition becoming a major research field within knowledge engineering.

The aim of knowledge acquisition is to develop methods and tools that make the arduous task of capturing and validating an expert's knowledge as efficient and effective as possible. Experts tend to be important and busy people; hence it is vital that the methods used, minimize the time each expert spends off the job taking part in knowledge acquisition sessions.

VII. KNOWLEDGE ENGINEERING PRINCIPLES

Since the mid-1980s, knowledge engineers have developed a number of principles, methods and tools that have considerably improved the process of knowledge acquisition. Some of the key principles are summarized as follows:

- Knowledge engineers acknowledge that there are different types of knowledge, and that the right approach and technique should be used for the knowledge required.
- Knowledge engineers acknowledge that there are different types of experts and expertise, such that methods should be chosen appropriately.
- Knowledge engineers recognize that there are different ways of representing knowledge, which can aid the acquisition, validation and re-use of knowledge.
- Knowledge engineers recognize that there are different ways of using knowledge, so that the acquisition process can be guided by the project aims.
- Knowledge engineers use structured methods to increase the efficiency of the acquisition process.

VIII. KNOWLEDGE ENGINEERING PROCESS:

The outline of Knowledge engineering process is given below:

1. Requirements analysis. Identify the scope of the knowledge-based system, typically in terms of its expected competency (for example, the kinds of queries it will be able to answer).
2. Conceptual modeling. Based on the scope defined in step 1, create a glossary of terminology (concepts) for the application domain and define interrelationships between the terms of and constraints on their usage. An explicit conceptual model of this kind is commonly called ontology.
3. Knowledge base construction. Using the conceptual model or ontology from step 2 as a collection of knowledge containers (or schemata), populate the knowledge base with instances of domain knowledge (often in the form of rules, facts, cases, or constraints).
4. Operationalization and validation. Operationalize the knowledge base from step 3 using automated reasoning mechanisms and validate its competence against the requirements from step 1. If satisfactory, release the system; otherwise, repeat steps 1 through 4 until satisfactory.
5. Refinement and maintenance. After delivery, the system continues to evolve as knowledge changes. Thus, steps 1 through 4 must be repeated throughout the life of the system.

Any knowledge management system that involves explicit knowledge representation is amenable to development using at least part of this process. In fact, it is always worth applying at least part of this process to any knowledge management activity that involves explicit knowledge representation.

IX. APPROACHES IN KNOWLEDGE ENGINEERING

A. Knowledge engineering as a Transfer Approach

The transfer process from human knowledge to knowledge base was implemented only with the development of Knowledge base. The required knowledge was obtained by interviewing experts on how they solve specific tasks and the same knowledge was implemented as production rules which were executed by an interpreter. The mixture of knowledge types, together with the lack of adequate justifications of the different rules, makes the maintenance of such knowledge bases very difficult and time consuming. Therefore, this transfer approach was only feasible for the development of small prototypical systems, but it failed to produce large, reliable and maintainable. knowledge bases.

Also, knowledge acquisition is the collection of already existing knowledge elements, was wrong due to the important role of tacit knowledge for an expert's problem-solving capabilities. These deficiencies resulted in a paradigm shift from the transfer approach to the modeling approach.

B. Knowledge engineering as a Modeling approach:

Building of KBS denotes modeling activity. It is like building a computer model with the aim of realizing problem-solving capabilities comparable to a domain expert and this model is created to offer similar solution in problem solving for problem in the area of concern (R. Studer et al., 1998). The term model implies that we have a representation or a formalism that mirrors behaviors in the real domain. Such model representation must provide semantics, implying that named elements within it correspond directly to named elements in the real domain (McCluskey 2002).

Following are the consequences of modeling view of Knowledge Based System:

1. Modeling approach is only an approximation of the reality. It is an infinite process with an aim to approximate the intended behaviour.
2. Modeling process is cyclic in nature. New observations may lead to a refinement, modification

or completion of the already built-up model. On the other side, the model may guide the further acquisition of knowledge.

3. The modeling process is dependent on the subjective interpretations of the knowledge engineer which may at times be faulty.

C. Specification Approach in Knowledge engineering:

Over the last 10 years a number of specification languages have been developed for describing KBSs. These specification languages can be used to specify the knowledge required by the system as well as the reasoning process which uses this knowledge to solve the task which is assigned to the system. On the one hand, these languages should enable a specification which abstracts from implementation details. On the other hand, they should enable a detailed and precise specification of a KBS at a level which is beyond the scope of specifications in natural language.

The essence of specification languages for KBSs

We identify three key features of specification languages for KBSs. First, most languages make use of a strong conceptual model to structure formal specifications. This reflects the fact that these languages were motivated by formalizing semiformal notations to describe KBSs. Therefore, these languages offer more than just a mathematical notation to define a computer program as an input-output relationship. Second, these languages have to provide means to specify the dynamic reasoning of a KBSs because this establishes a significant piece of the expertise required by such systems. Third, a KBS uses a large body of knowledge requiring structured and rich primitives for representing it.

Representing rich knowledge structures

Most of the specification languages provide at least epistemological primitives like constants, sorts/types, functions, predicates/relations and some mathematical toolkit as a means of specifying the static aspects of a system. However, richer languages provide additional syntactical sugar on top of these mathematical primitives. In particular, object-oriented or frame-based specification languages provide a rich variety of appropriate modelling primitives for expressing static system aspects: values, objects, classes, attributes with domain and range restrictions, set-valued attributes, is-a relationships with attribute inheritance, aggregation, grouping etc. For example, KARL provides such modelling primitives which enables a smooth transformation from semiformal specification languages like CML to formal specifications of these models.

In the following, we briefly sketch three different specification languages for KBSs: (ML) 2, KARL and DESIRE. A more detailed discussion and comparison can be found in (ML) 1 which was developed as part of the KADS projects, is a formalization language for KADS Expertise Models. The language provides a formal specification language for the KADS Expertise Model by combining three types of logic: order-sorted first-order logic extended by modularization for specifying the domain layer, first-order meta-logic for specifying the inference layer, and quantified dynamic logic for specifying the task layer. KARL [53] is an operational language which restricts the expressive power of the object logic by using a variant of Horn logic. It was developed as part of the MIKE project and provides a formal and executable specification language for the KADS Expertise Model by combining two types of logic: L-KARL and P-KARL. L-KARL, a variant of Frame Logic, is provided to specify domain and inference layers. It combines first-order logic with semantic data-modelling primitives. A restricted version of dynamic logic is provided by P-KARL to specify a task layer. Executability is achieved by restricting Frame logic to Horn logic with stratified negation and by restricting dynamic logic to regular and deterministic programs.

The language DESIRE relies on a different conceptual model for describing a KBS: the notion of a compositional architecture. A KBS is decomposed into several interacting components. Each component contains a piece of knowledge at its object-layer and has its own control defined at its internal meta-layer. The interaction between components is represented by transactions and the control flow between these modules is defined by a set of control rules. DESIRE extensively uses object-meta relationships to structure specifications of KBSs. At the object-level, the system reasons about the world state. Knowledge about how to use this knowledge to guide the reasoning process is specified at the meta-level. The meta-level reasons about controlling the use of the knowledge specified at the object-level during the reasoning process. The meta-level describes the dynamic aspects of the object-level in a declarative fashion. A module may reason on its object-level about the meta-level of another module.

From a semantic point of view a significant difference between DESIRE on the one hand and (ML) 2 and KARL on the other hand lies in the fact that the former uses temporal logics for specifying the dynamic reasoning process whereas the latter use dynamic logic. In dynamic logic, the semantics of the overall program is a binary relation between its input and output sets. In DESIRE, the entire reasoning trace which leads to the derived output is used as semantics.

X. KNOWLEDGE ENGINEERING METHODOLOGIES

The important methodologies followed while developing Knowledge engineering systems are as follows:

1. CommonKADS

2. MIKE approach

3. PROTE'GE'- II approach

4. SPEDE

5. MOKA

A. CommonKADS: It is a complete methodological framework for the development of a knowledge based system (KBS). It supports most aspects of a KBS development project, such as:

- Project management
- Organisational analysis (including problem/opportunity identification)
- Knowledge acquisition (including initial project scoping)
- Knowledge analysis and modelling
- Capture of user requirements
- Analysis of system integration issues
- Knowledge system design

CommonKADS describes KBS development from two perspectives:

1. **Result perspective**: A set of models, of different aspects of the KBS and its environment that are continuously improved during a project life-cycle.
2. **Project management perspective**: A risk-driven generic spiral life-cycle model that can be configured into a process adapted to the particular project.

B. MIKE Approach:

The MIKE approach (Model-based and Incremental Knowledge Engineering) provides a development method for KBSs covering all steps from the initial elicitation through specification to design and implementation. MIKE proposes the integration of semiformal and formal specification techniques and prototyping into an engineering framework. Integrating prototyping and support for an incremental and reversible system development process into a model-based framework is actually the main distinction between MIKE and CommonKADS

MIKE takes the Expertise Model of CommonKADS as its general model pattern and provides a smooth transition from a semiformal representation, the Structure Model, to a formal representation, the KARL Model, and further to an implementation oriented representation, the Design Model. The smooth transition between the different representation levels of the Expertise Model is essential for enabling incremental and reversible system development in practice.

In MIKE the executability of the KARL Model enables validation of the Expertise Model by prototyping. This considerably enhances the integration of the expert in the development process. The entire development process is divided into a number of subactivities: Elicitation, Interpretation, Formalization/Operationalization, Design and Implementation. Each of these activities deals with different aspects of the system development.

C. PROTE'GE'- II Approach

The PROTE'GE'-II approach aims at supporting the development of KBSs by the reuse of PSMs (Problem solving methods) and ontologies. In addition, PROTE'GE'-II puts emphasis on the generation of custom tailored knowledge-acquisition tools from ontologies. PROTE'GE'-II relies on the task-method-decomposition structure (compare Section 2.2): By applying a PSM a task is decomposed into corresponding subtasks. This decomposition structure is refined down to a level at which primitive methods, so called mechanisms, are available for solving the subtasks. Up to now, PROTE'GE'-II offers a small library of implemented PSMs which have been used to solve a variety of tasks.

In PROTE'GE-II the input and output of a method is specified by a so called method ontology : such a method ontology defines the concepts and relationships that are used by the PSM for providing its functionality. For example, the Board-Game Method uses among others the notions of 'pieces', 'locations' and 'moves' to provide its functionality, that is, to move pieces between locations on a board. In this way a method ontology corresponds to the generic terminology as introduced by the collection of knowledge roles of a PSM. A second type of ontology used within PROTE'GE'-II are domain ontologies: they define a shared conceptualization of a domain. Both PSMs and domain ontologies are reusable components for building up a KBS. However, due to the interaction problem the interdependence between domain 0ntologies and PSMs with their associated method ontologies has to be taken into account when constructing a KBS from reusable components. Therefore, PROTE'GE'-II proposes the notion of an application ontology to extend domain ontologies with PSM specific concepts and relationships.

D. SPEDE

The SPEDE methodology is a combination of principles, techniques and tools taken from Knowledge Engineering and adapted for use in Knowledge Management. It provides an effective means to capture, validate and communicate vital knowledge to provide business benefit.

There are organizations such as Rolls-Royce who has run over 100 SPEDE projects, involving the training of over 150 employees.

i. Structure and Deliverables

SPEDE has been specifically developed to act as a training course for novice knowledge engineers or those seconded to a knowledge management activity. SPEDE projects typically involve 1-week of intensive training followed by 2-3 months of scoping, knowledge acquisition and delivery phases.

The main deliverable of most SPEDE projects is an intranet website. However, previous projects have delivered quality procedures, process improvement information and expert systems.

Projects using the SPEDE methodology follow a set of procedures coordinated by experienced staff. All projects have a coach who manages the activities of one or more knowledge engineers on a daily basis.

ii. Gates

All SPEDE projects must pass through a series of gates. These are meetings held at various stages throughout the project to act as a "go/no go" into the next phase of the project. Each gate comprises various criteria to ensure the project is on track to meet the objectives and identify any problems, hazards and actions. There are 5 gates: project launch review, scoping review, technical review, delivery review and post-delivery review.

E. MOKA: Methodology and tools Oriented to Knowledge-Based Engineering Applications

MOKA is a methodology for developing Knowledge-Based Engineering applications, i.e. systems that support design engineers. It is particularly aimed at capturing and applying knowledge within aeronautical and automotive industries of the design of complex mechanical products.

Whilst huge benefits can be gained by the use of Knowledge-Based Engineering (KBE) technology, the lack of a recognised methodology has resulted in a significant risk when developing and maintaining KBE applications. MOKA aims to provide such a methodology, that:

- Reduces the lead times and associated costs of developing KBE applications by 20 - 25%.
- Provides a consistent way of developing and maintaining KBE applications.
- Will form the basis of an international standard.
- Makes use of a software tool to support the use of the methodology.

i. Need for MOKA

Companies have to manage and reuse engineering knowledge to improve business processes, to reduce time to find new solutions, to make *correct first time* and to retain best practices. The aim of MOKA is to provide a methodology to capture and formalise engineering knowledge to reuse it, for example within KBE applications. Development and maintenance of knowledge intensive software applications is a complex and potentially expensive activity. The number of Knowledge-Based Engineering (KBE) systems used in the aeronautical and automotive industries has increased in recent years. Experience has shown that long term risk can be reduced by employing a systematic methodology that covers the development and maintenance of such systems.

ii. MOKA Analysis and Modeling

MOKA identifies two models to be used in the KBE application development lifecycle :

- **Informal Model**: A structured, natural language representation of engineering knowledge using pre-defined forms.
- **Formal Model**: A graphical, object-oriented representation of engineering knowledge at one level of abstraction above application code.

Within each of these models, various knowledge representations are used to help capture, analyse and structure the knowledge required for KBE applications.

Within the informal model, the main knowledge objects are:

- **Entities**
  - **Structural Entities** (the components of the product being designed)
  - **Functional Entities** (the functions of the product and its sub-components)
- **Constraints** (the design requirements of the product and its sub-components)
- **Activities** (the tasks performed during the design process)
- **Rules** (decision points in the design process that affect what tasks to perform)
- **Illustrations** (examples that illustrate aspects of the product and design)

iii. MOKA Tool

PCPACK can be used to satisfy the requirements for a supporting software tool for the MOKA methodology. It supports the capture, analysis, modelling and publishing of design knowledge using a MOKA framework.

MOKA consistes of the following partners: Aerospatiale Matra (prime), British Aerospace, Daimler-Chrysler, PSA Peugeot Citroen, Knowledge Technologies International, Decan and Coventry University. A MOKA Interest Group continues to meet and develop the methodology.

XI. KNOWLEDGE ENGINEERING TRENDS

Some of the trends in Knowledge Engineering in the last few years are discussed in this section.The text below is a brief overview of paper "Knowledge Engineering: Principles and methods" (Rudi Studer a'*, V. Richard Benjamins b'c, Dieter Fensel, 1998).

a. The paradigm Shift from a transfer view to a modeling view

According to the transfer view the human knowledge required to solve a problem is transferred and implemented into the knowledge base. However this assumes that concrete knowledge is already present in humans to solve a problem. The transfer view disregards the tacit knowledge an individual acquires in order to solve a problem. This is one of the reasons for a paradigm shift towards modeling view. This shift is compared to a shift from first generation expert systems to second generation expert systems.

The modeling view is a closer approximate of reality and perceives solving problems as a dynamic, cyclic, incessant process dependent on the knowledge acquired and the interpretations made by the system. This is similar to how an expert solves problems in real life.

b. The evolving of Role Limiting methods and Generic Tasks

Role limiting methods are based on reusable problem solving methods. Different knowledge roles are decided and the knowledge expected from each of these roles is clarified. However the disadvantage of role limiting methods is that there is no logical means of deciding whether a specific problem can be solved by a specific role-limiting method.

This disadvantage gave rise to Configurable role limiting methods. Configurable role limiting methods are based on the idea that a problem solving method can further be broken up into several smaller sub tasks each task solved by its own problem solving method.

Generic Tasks include a rigid knowledge structure, a standard strategy to solve problems, a specific input and a specific output.

The GT approach is based on the strong interaction problem hypothesis which states that the structure and representation of domain knowledge is completely determined by its use

c. The usage of Modeling Frameworks

The development of Specification languages and problem solving methods of knowledge based systems. Over the past few years the modeling frameworks that became prominent within Knowledge engineering are Common KADS, MIKE (Model-based and Incremental knowledge engineering) and PROTÉGÉ-II is a modeling framework influenced by the concept of ‘Ontology’.

d. The influence of Problem solving methods and Ontology

Ontologies help building model of a domain and define the terms inside the domain and the relationships between them. There are different types of Ontologies including Domain ontologies, Generic ontologies, application ontologies and representational ontologies.

While categorizing knowledge, storing, retrieving and managing information is not only useful for solving problems without direct need of human expertise but also leads to ‘Knowledge Management’ efforts that enable an organization to function efficiently in the long run.

XII. RELATIONSHIP BETWEEN KNOWLEDGE MANAGEMENT AND KNOWLEDGE ENGINEERING

Knowledge engineering deals with the development of information systems in which knowledge and reasoning play important roles. Knowledge management is a newly developed field with an intersection of computer science and management and deals with “knowledge” as a key resource in modern organizations. Managing knowledge within an organization is inconceivable without the use of advanced information systems; the design and implementation of such systems create great organization as well as technical challenges.

Knowledge Management and Knowledge engineering can be used interchangeably just like the terms data and information. The term “management” points to managing executive, administrative and supervisory direction, where as, “engineering” is to lay out, construct, contrive or plan out, usually with more or less with greater competencies or skill level.

So, we can state that a knowledge manager establishes direction to the process and the knowledge engineer develops the means to accomplish that direction. A Knowledge manager should focus on the knowledge needs of the organization. They should update themselves doing latest research to understand the kind of knowledge required to make certain decision and the actions that need to be taken. Knowledge Managers should play a pivotal role in the design of the enterprise starting from the needs of the enterprise to establishing the enterprise level knowledge management policies. It is to the knowledge managers that the user should go with their "need to know".

Regarding Knowledge Engineers, they work on areas such as data and information, representation and encoding methodologies, data repositories, work flow management, groupware technologies, etc. The knowledge engineers carry out research on the technologies needed to meet the enterprise's knowledge management needs, establish the processes by which knowledge requests are examined, information assembled, and knowledge returned to the requestor.

There may be other views pertaining to the roles of a Knowledge Manager and the Knowledge Engineer. For example, to the developer of knowledge-base computer software systems, the knowledge engineer is most likely a computer scientist specializing the development of artificial intelligence knowledge bases. From the view of the corporate board-room the knowledge manager may be the Chief Information Officer (CIO) or the person in charge of the Information Resource Management (IRM). The point is, when discussing terms such as knowledge manager or knowledge engineer, or any other role designation, it is important that all parties should share a clear mutual understanding. It is very true that Knowledge Engineering and Knowledge Management is concerned with all aspects of eliciting, acquiring, modeling and managing knowledge, and their role lies in the effective construction of knowledge-intensive systems and services for the semantic web, knowledge management, e-business, natural language processing, intelligent integration information, etc.

XIII. CURRENT MANAGEMENT PRACTICES:

Most knowledge management activities combine business processes and engineering technology. As currently practiced, knowledge management includes several activities and technologies:

- Document management systems allow workers to find existing documents relevant to the task at hand. Essentially, these are multisource search and information- retrieval systems that tie into an organization's intranet (and may extend to the public Internet). These systems include several commercially available products, such as those made by Autonomy and Verity.
- Discussion forum systems promote knowledge dissemination within communities of practice. Workers subscribe to forums relevant to their interests, exchanging questions and answers, lessons learned, announcements, and industry gossip. Such systems are easily implementable with both freely available Web software and commercial products.
- Capability management systems allow an organization to "know who knows what." Essentially, these are databases of suitably structured CVs or resumes; as such, they are implementable with off-the-shelf database software. The goal is to put people together by matching one person's need for expertise with another person's listed skills.
- Lessons-learned knowledge base systems let workers tap into past experience, by storing that experience as structured cases. These systems allow sophisticated queries, typically supporting "fuzzy" retrieval of "similar" cases. Although simple systems can use just conventional database software, full functionality requires special-purpose, case-based reasoning or knowledge-based system software.

XIV. CONCLUSION:

Knowledge can be managed through Knowledge engineering techniques for which Knowledge Management activities require a systematic framework. When there lies a strong bond between Knowledge Engineering and Knowledge Management, various knowledge processes such as modeling, verification, storage, querying, and updating can be integrated. Again, continuous change of knowledge in knowledge process can be supported. Further, with hierarchical ontology system, knowledge sharing and knowledge interoperations can be also be supported. As current business scenario is highly dynamic and competitive, Knowledge engineering and Knowledge Management requires greater integration and co-ordination.

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Low Complexity Full Rate Soft Detection of Terrestrial Digital TV Broadcasting

Eugin E and Jeno Paul P

***Abstract*—Last-generation and future wireless communication standards, such as DVB-T2 or DVB-NGH, are including multi-antenna transmission and reception in order to increase bandwidth efficiency and receiver robustness. The main goal is to combine diversity and spatial multiplexing in order to fully exploit the multiple-input multiple output (MIMO) channel capacity. Full-rate full-diversity (FRFD) space-time codes (STC) such as the Golden code are studied for that purpose. However, despite their larger achievable capacity, most of them present high complexity for soft detection, which hinders their combination with soft-input decoders in bit-interleaved coded modulation (BICM) schemes. This article presents a low complexity soft detection algorithm for the reception of FRFD space-frequency block codes in BICM orthogonal frequency division multiplexing (OFDM) systems. The proposed detector maintains a reduced and fixed complexity, avoiding the variable nature of the list sphere decoder (LSD) due to its dependence on the noise and channel conditions. Complexity and simulation based performance results are provided which show that the proposed detector performs close to the optimal log-maximum a posteriori (MAP) detection in a variety of DVB-T2 broadcasting scenarios.**

Index Terms—MAP detection, MIMO systems, space frequency block coding (SFBC), low-density parity check (LDPC) codes, digital video broadcasting(DVB), orthogonal frequency division multiplexing (OFDM).

I. INTRODUCTION

SPACE-TIME coding is one of the main methods in order to exploit the capacity of multiple-input multiple-output (MIMO) channels [1]. Since STC techniques use both time and spatial domains for coding data symbols, diversity and spatial multiplexing can be combined achieving robustness at the receiver with a higher data rate transmission. As a result, STC techniques have been incorporated in many of the last-generation wireless communications systems, including the new generation of terrestrial and mobile digital video broadcasting (DVB) standards. If STC is joined to multi-carrier modulation, such as orthogonal frequency-division multiplexing (OFDM), spacefrequency block coding (SFBC) can be performed. This way, codewords are fed into adjacent of the two consecutive OFDM symbols, translated to the time domain and transmitted through several transmit antennas. This transmission scheme is usually combined with bit-interleaved coded modulation (BICM) giving good diversity results in a wireless communication link [2].

In order to achieve the full MIMO diversity-multiplexing frontier [3], the proposals for the future generations of terrestrial, portable and mobile digital video broadcasting standards, such as DVB-NGH, focus on the combination of both diversity and spatial multiplexing [4], [5] through full-rate full-diversity (FRFD) codes [6]. The main drawback of full-rate codes arises from their very high decoding complexity, which grows exponentially with the number of transmitted symbols per codeword. In order to reduce the complexity of the detection process, hard detection techniques such as sphere decoding (SD) or low complexity STC designs [7], [8] can be used. Nevertheless, when iterative decoders, such as turbo or LDPC codes, are included in the reception chain, soft information on the conditional probabilities for all possible transmitted symbols is required in the form of log-likelihood ratios (LLR). Moreover, the computation of the LLRs for the whole set of transmitted symbols is unfeasible, specially for large constellation sizes. Hence, the soft MIMO detector has to select a group of candidates to be fed to the decoder in order to compute the required LLRs. Several algorithms that serve this purpose can be found in the literature, such as list sphere detection (LSD) [9], near-optimal soft SD [10], list fixed-complexity sphere detection (LFSD) [11], QR decomposition associated with the M-algorithm (QRM-MLD) [12] or bounded soft sphere detection (BSSD) [13].

The contribution of this paper is a new low-complexity soft detection algorithm for FRFD SFBC and its assessment in an LDPC-based BICM scenario. The generation of the candidate list is carried out by means of a fixed tree search, whose complexity does not depend on the channel conditions or the noise level. Since the complexity of the proposed soft detector is closely linked to the architecture of the tree search, we analyze different tree configurations in order to find the best balance between complexity and performance. Simulation results are provided for the Golden [14] and the FRFD Sezginer-Sari (SS) codes [8] in a DVB-T2 framework [15]. Although our research has been carried out on a terrestrial **TV** system, the results can be generalized for any MIMO bitinterleaved coded modulation scheme. The remainder of the paper is organized as follows: Section II details the system model, focusing on soft detection and LLR calculation. The design of the new algorithm is presented for two FRFD SFBCs in Section III, while Section IV shows the simulation-based performance of the proposed receiver in DVB-T2 scenarios. Finally, the main concluding remarks are drawn in Section V.

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II. SYSTEM MODEL

The basic structure of the LDPC-coded BICM-OFDM system is depicted in Fig. 1. As can be seen, the bit stream is coded, interleaved and mapped onto a complex constellation. Next, a vector of Q symbols s is coded into

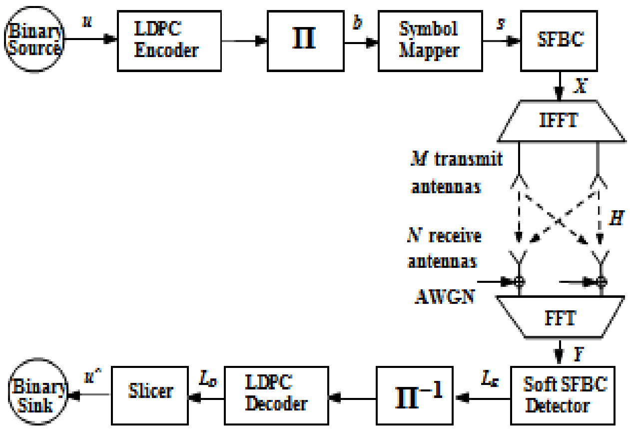

Fig. 1. Simplified diagram of a LDPC-based SFBC MIMO transmission and reception scheme based on DVB-T2.

space and frequency forming the codeword **X**, which is transformed into the time domain by an inverse fast fourier transform (IFFT) block and transmitted after the addition of fast Fourier transform (FFT) is carried out and the resulting signal **Y** of dimensions $N\times T$ can be represented mathematically as

$$\boldsymbol{Y} = \boldsymbol{HX} + \boldsymbol{Z}, \qquad (1)$$

where H denotes the $N\times M$ complex channel matrix, **X** is any $M\times T$ codeword matrix composed by a linear combination of Q data symbols and **Z** represents the $N\times T$ zero-mean additive white Gaussian noise (AWGN) matrix whose complex coefficients fulfill $\mathcal{CN}$ $(0, 2\sigma^2)$ being σ^2 the noise variance per real component. The design of the codeword **X** follows the criteria defined in [16], [17] and will provide full rate when $Q = MT$, being T the frequency depth of the codeword. By taking the elements column-wise from matrices **X, Y** and **Z**, equation (1) can be vectorized as

$$\boldsymbol{y} = \check{H}\boldsymbol{Gs} + \boldsymbol{z}, \qquad (2)$$

where y, s and z are column vectors. The matrix $\check{H}$ is the equivalent $NT\times MT$ MIMO channel written as

$$\check{H} = \begin{bmatrix} H^1 & 0 & \cdots & 0 \\ 0 & H^2 & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & H^T \end{bmatrix} \qquad (3)$$

where we have a block diagonal of channel realizations $\mathbf{H}^c$ at the carriers $c = 1, \ldots, T$. The complex coefficient h^c_{ij} corresponds to the channel from transmit antenna j to receive antenna i at the c-th carrier. The off-diagonal entries are zero matrices with dimensions $N\times M$. The matrix **G** is the generator matrix for the SFBC such that x = **G**s, where s corresponds to the symbol column vector $[s1, \ldots, sQ]^T$.

A. *Soft Detection of SFBCs*

The soft information required by the LDPC decoder is obtained though maximum a posteriori (MAP) detection, which consists of evaluating the LLR of the*a posteriori* probabilities of a bit bk taking its two possible values, i.e.$L_D(b_k) = \ln\frac{\Pr[b_k=+|y]}{\Pr[b_k=-|y]}$ with $k = \{0,\ldots, MT\log_2 P-1\}$. Assuming statistically independent information bits and using the Bayes' rule, the a posteriori information $L_D(b_k|\boldsymbol{y})$ can be expressed as $L_D(b_k|\boldsymbol{y}) = L_A(b_k)+L_E(b_k|\boldsymbol{y})$, where $L_A(b_k)$ and $L_E(b_k|\boldsymbol{y})$ denote the a priori and extrinsic information, respectively. Considering our non-iterative model depicted in Fig. 1 and the vectorized model (2), the extrinsic information can be written using the Max-log approximation as

$$L_E(b_k/y) \approx \frac{1}{2}\max_{b\in B_{k,+1}}\left\{-\frac{1}{\sigma^2}\|y-\check{H}G_s\|^2\right\} - \frac{1}{2}\max_{b\in B_{k,-1}}\left\{-\frac{1}{\sigma^2}\|y-\check{H}G_s\|^2\right\}, \qquad (4)$$

Where $\mathbb{B}_{k,\pm1}$ represents the sets of $2^{MT\log2 P-1}$ bit vectors $\boldsymbol{b}$ having $b_k = \pm1$ and the symbol column vector $s = map(\boldsymbol{b})$ is the mapping of the vector $\boldsymbol{b}$ into the symbols o column vector $\boldsymbol{s}$. The main difficulty in the calculation of (4) arises from the computation of the metrics $\|\boldsymbol{y}- \check{\boldsymbol{H}}\boldsymbol{Gs}\|^2$ since a calculation of P^Q metrics is necessary for a FRFD SFBC, being P the modulation order. This becomes unfeasible for high modulation orders unless the calculation of (4) can be reduced. As a result, a good approximation based on a candidate list $\mathcal{L}$ is proposed in [9] in order to reduce the calculation of L_E in (4). The list includes $1 \leq N_{cand} < P^Q$ vectors s with the smallest metrics and the number of candidates N_{cand} must be defined sufficiently large in such a way that it contains the maximizer of (4) with high probability. Hence, (4) can be approximated as

$$L_E(b_k/y) \approx \frac{1}{2}\max_{b\in\mathcal{L}\cap B_{k,+1}}\left\{-\frac{1}{\sigma^2}\|y-\check{H}G_s\|^2\right\} - \frac{1}{2}\max_{b\in\mathcal{L}\cap B_{k,-1}}\left\{-\frac{1}{\sigma^2}\|y-\check{H}G_s\|^2\right\}, \qquad (5)$$

III. FIXED-COMPLEXITY DETECTION

The design of efficient detection algorithms is one of the greatest challenges when implementing full-rate SFBC. Given the high complexity of performing an exhaustive search, special focus has been drawn into developing lower complexity detection algorithms that yield a close-to-ML performance. The LSD is one of the most remarkable approach but its complexity order is bounded by $\mathcal{O}(P^Q)$ in the same way as the SD [9]. Even though the list of candidates corresponds to the set $\mathcal{L}$ of the smallest metrics, the complexity of performing such a selection may be considerably high for low signal-tonoise ratio (SNR) scenarios. Furthermore, an unsuitable choice of the initial radius may lead to a shortage in candidate points, which forces the algorithm to restart with a looser sphere onstraint. In order to limit the complexity and to facilitate the computation of soft detected symbols, a fixed-complexity tree-searchstyle algorithm was proposed in [18] for spatial multiplexing schemes, coined list fixed-complexity sphere decoder (LFSD).

The main feature of the LFSD is that, instead of constraining the search to those nodes whose accumulated Euclidean distances are within a certain radius from the received signal, the search is performed in an unconstrained fashion. The tree search is defined instead by a tree configuration vector $n = [n_1, \ldots, n_{MT}]$, which determines the number of child nodes (n_i) to be considered at each level. Therefore, the tree is traversed level by level regardless of the sphere constraints. Once the bottom of the

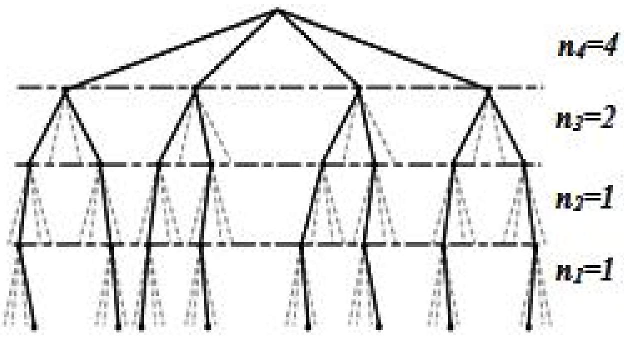

Fig. 2. Fixed-complexity tree search of a QPSK-modulated signal using a tree configuration vector of n = [1, 1, 2, 4].

tree is reached, the detector retrieves a list of N_{cand} candidate symbol vectors. It is worth noting that the set G composed the vectors of the set $\mathcal{L}$ with the smallest metrics given by the LSD, but provides sufficiently small metrics and diversity of bit values to obtain accurate soft information. A representation of an LFSD tree search is depicted in Figure 2 for a QPSK modulation and a tree configuration vector of n = [1, 1, 2, 4].

A. Soft-output LFSD algorithm

For the sake of simplicity, the equations for the aforementioned FRFD codes will be rearranged so that ML metrics can be given by $\|\bar{y} - \boldsymbol{H}_{eq\bar{s}}\|^2$, where $\bar{y}$ and $\bar{s}$ are the received and transmitted signals respectively, reorganized according to the Corresponding SFBC code, and H_{eq} is the effective equivalent channel, which will be defined later for the SS and Golden codes.

A level-by-level computation of the metrics requires the conversion to the following equivalent system

$$\|U(\bar{s} - \hat{\bar{s}})\|^2, \tag{6}$$

where $\boldsymbol{U}$ is obtained through the Cholesky decomposition of $\mathbf{H}^{H}_{eq}\mathbf{H}_{eq}$ and $\hat{\bar{s}} = \mathbf{H}^{\dagger}_{eq}\,\bar{y}$. Given the triangular structure of $\boldsymbol{U}$, it is now possible to compute the accumulated Euclidean distances (AED) up to level i recursively by traversing the tree backwards from level $i = MT$ down to $i = 1$. The Euclidean distances that must be minimized in the cost function in (6) can be equivalently represented in a tree search fashion as

$$D_i = u_{ii}^2|\bar{s}_i - Z_i|^2 + \sum_{j=i+1}^{MT} u_{jj}^2|\bar{s}_j - Z_j|^2 = d_i + D_{i+1}, \tag{7}$$

and

$$Z_i = \hat{\bar{s}}_i - \sum_{j=i+1}^{MT} \frac{u_{ij}}{u_{ii}}\left(\bar{s}_j - \hat{\bar{s}}_j\right). \tag{8}$$

Therefore, the n_i symbols to be evaluated at each level i are chosen in accordance with the Schnorr-Euchner enumeration [19], being their corresponding partial Euclidean distances d_i computed and accumulated to the previous level's AED, that is, D_{i+1}. Once the bottom of the tree has been reached, a sorting operation is performed on the $n_T = \prod_{i=1}^{MT} n_i$ Euclidean distances in order to select the N_{cand} symbol vectors with the smallest metrics. This latter sorting procedure can be avoided if the tree configuration vector n is chosen so as to yield $n_T = N_{cand}$. In such a case, the complexity of the algorithm is reduced at the expense of a degradation in the quality of soft information as the selected metrics are higher in value.

B. Ordering algorithm

The performance of the LFSD soft-detector in uncoded scenarios is strongly dependent on the ordering algorithm of the channel matrix and the choice of the tree configuration vector [18]. The ordering algorithm proposed in [18] to enhance the performance of the LFSD and FSD detectors was based on the fact that it was possible to mitigate the error propagation derived from ruling out several tree branches by ordering the several columns of the channel matrix according to their *quality*.

More precisely, the FSD ordering scheme dictates that the subchannel with the worst norm needs to be processed first, since all the constelation symbols are evaluated at the first level (see Fig. 2), and therefore, there is no error propagation to the remainder levels. However, in the specific case of spacefrequency- coded systems, the effect of the ordering algorithm on the overall performance relies on the type of code utilized. In order to verify this assumption, two 2 × 2 FRFD SFBC codes have been assessed.

1) Golden code [14]: For the Golden code, the data symbol vector s is transformed into the transmitted codeword as follows:

$$\mathbf{X_g} = \frac{1}{\sqrt{5}}\begin{bmatrix} \alpha(s_1 + \theta s_3) & \alpha(s_2 + \theta s_4) \\ i\bar{\alpha}(s_2 + \bar{\theta} s_4) & \bar{\alpha}(s_1 + \bar{\theta} s_3) \end{bmatrix}, \tag{9}$$

With $\theta = \frac{1+\sqrt{5}}{2}$ (the golden number), $\bar{\theta} = \frac{1-\sqrt{5}}{2}$,

$\alpha = 1 + i - i\theta$ and $\bar{\alpha} = 1 + i - i\bar{\theta}$.

The vectorization of (1) with $\boldsymbol{Xg}$ implies the following equivalent channel

$\boldsymbol{H_{eq}}$= (10)

$$\begin{bmatrix} h^1_{11}(1+i\bar{\theta}) & h^1_{12}(-\theta+i) & h^1_{11}(\theta+i) & h^1_{12}(-1+i\bar{\theta}) \\ h^1_{21}(1+i\bar{\theta}) & h^1_{22}(-\theta+i) & h^1_{21}(\theta+i) & h^1_{22}(-1+i\bar{\theta}) \\ h^2_{12}(1+i\theta) & h^2_{11}(1+i\bar{\theta}) & h^2_{12}(\bar{\theta}+i) & h^2_{11}(\theta+i) \\ h^2_{22}(1+i\theta) & h^2_{21}(1+i\bar{\theta}) & h^2_{22}(\bar{\theta}+i) & h^2_{21}(\theta+i) \end{bmatrix}$$

The fact that Golden code does not equally disperse the symbol energy in all spatial and temporal directions, generates an unbalanced structure of the transmitted symbols since the norms of the code weights in (10) are not equal, i.e. $1+\theta^2 \neq 1+\bar{\theta}^2$. Thus, given this difference in the absolute value of the weights, one of the symbols in each transmitted pair (s_1, s_3) and (s_2, s_4) always has a higher power than the other. Hence, the norms of the equivalent subchannels are $\|h_1\|^2 \neq \|h_3\|^2$ and $\|h_2\|^2 \neq \|h_4\|^2$ in any case, being h_j the j-th column vector of $\boldsymbol{H}_{eq}$. This unbalanced structure allows for the implementation of a new ordering procedure in order to improve the overall system's performance.

On the other hand, if the channel is assumed quasi-static over adjacent carriers, are $\|\mathbf{h_1}\|^2 \approx \|\mathbf{h_3}\|^2$ and $\|\mathbf{h_2}\|^2 \approx \|\mathbf{h_4}\|^2$. An important feature when considering the optimum ordering approach is the tree configuration vector that will shape the search tree. As opposed to the LFSD detector presented in [18] for spatial multiplexing MIMO transmission, the tree configuration vector for the detection of the Golden Code has been set to $n = [k, k, P, P]$, where $k < P$. With such a tree structure, an exact ML search is performed in the first two levels of the tree, and therefore, there is no error propagation down to the next levels.

Consequently, by ordering the equivalent channel matrix in such a way that the *worst* subchannel is processed in the first two levels of the tree, the probability of finding vectors with smaller metrics is increased. Moreover, it has to be taken into account that the symbols belonging to the same pair need to be detected together in the non-ML part of the tree search for a better performance of the algorithm, since there exists a correlation between their corresponding subchannels due to the code structure.

The equivalent ordered channel matrix $\boldsymbol{H}_{eq}^{ord}$, which will be used in the detection of the Golden Code, can then be described as

$$\boldsymbol{H}_{eq}^{ord} = [h_{\tilde{b}}\ h_b\ h_{\tilde{w}}\ h_w], \tag{11}$$

Where

$$w = argmin_{j\in S}\ \|h_j\|^2, \tag{12}$$

and

$$b = argmax_{j\in S, j\neq w,\tilde{w}}\ \|h_j\|^2. \tag{13}$$

The two symbols that compose a symbol pair are represented as $(a, \bar{a})$ and the set of symbol indices is $S = \{1,\ldots,MT\}$. Given the chosen tree configuration vector, one can notice that the order of the first two selected symbols can be switched without having any impact on the final performance of the system.

2) The SS code: This 2×2 SFBC scheme was designed to enable optimum detection with lower complexity than the Golden code. The low complexity detection property of multistrata codes such as the SS code, was analyzed in [7], [8],where it was proven that optimum output was obtained with two symbol-by-symbol detection stages of complexity P and a ML detection of complexity $\mathcal{O}(P^2)$. The main difference between the SS [8] and the silver codes [7] is the larger coding gain of the latter. The SS code is the combination of two Alamouti schemes whose codeword can be written as

$$\boldsymbol{X}_{ss} = \begin{bmatrix} as_1 + bs_3 & as_2 + bs_4 \\ -cs^*_2 - ds^*_4 & cs^*_1 - ds^*_3 \end{bmatrix}, \tag{14}$$

With $a = c = 1/\sqrt{2}$, $b = \frac{1-\sqrt{7}+i(1+\sqrt{7})}{4\sqrt{2}}$ and $d = -ib$.

The vectorization of (1) for $\boldsymbol{X}_{ss}$ implies a rearrangement of the received and transmitted signals such that $\bar{\mathbf{y}} = [y_{11}\, y_{21}\, y^*_{12}\, y^*_{22}]^T$ and $\bar{\mathbf{s}} = [s_1, s_2, s^*_3, s^*_4]^T$ respectively.

The equivalent channel can be expressed as:

$$\boldsymbol{H}_{eq} = \begin{bmatrix} ah^1_{11} & -ch^1_{12} & bh^1_{11} & -dh^1_{12} \\ ah^1_{21} & -ch^1_{22} & bh^1_{21} & -dh^1_{22} \\ (ch^2_{12})^* & (ah^2_{11})^* & (dh^2_{12})^* & (bh^2_{11})^* \\ (ch^2_{12})^* & (ah^2_{21})^* & (dh^2_{22})^* & (bh^2_{21})^* \end{bmatrix}, \tag{15}$$

When considering the equivalent channel in (15), it is worth noting that the equivalent subchannels for the symbol pairs ($s1$, $s3$) and ($s2$, $s4$) have very similar norms. This is due to two main factors. On one hand, both symbol pairs undergo almost the same channel conditions as they are assigned to adjacent carriers ($\boldsymbol{H}^1 \approx \boldsymbol{H}^2$) in quasi-static fading channels). On the other hand, the code weights a, b, c and d imposed by the SS code fulfill a power constraint for linear dispersion codes [20], which forces the symbols to be dispersed with equal energy in all spatial and temporal directions, i.e. $|a| = |b| = |c| = |d| = 1/\sqrt{2}$.

The consequence of employing such a code is that the difference of the norms of the equivalent subchannels is negligible and, therefore, performing a matrix ordering stage does not provide any remarkable performance enhancement. As it will be shown in the next section, the proposed ordering approach yields close-to-optimum performance when combined with the suggested tree configuration vector. Moreover, the matrix ordering process only requires the computation of MT vector norms as opposed to other ordering algorithms such as FSD [18] or V-BLAST [21], which need to perform $MT - 1$ matrix inversion operations.

C. Bit LLR generation for the proposed list fixed-complexity detector

The expression for the LLRs in (5) can be rewritten to comply with the equivalent system in (6) as

$$L_E\left(b_{k|\mathbf{y}}\right) \approx \frac{1}{2}\min_{b\in\mathfrak{G}\cap B_{k,+1}} \frac{1}{\sigma^2}\left\|\bar{y}-\boldsymbol{H}_{eq}\bar{s}\right\|^2 - \frac{1}{2}\min_{b\in\mathfrak{G}\cap B_{k,-1}} \frac{1}{\sigma^2}\left\|\bar{y}-\boldsymbol{H}_{eq}\bar{s}\right\|^2, \qquad (16)$$

Note that the list of candidates $\mathcal{L}$ of the ML/LSD detector has been substituted by the set G of the LFSD.

IV. SIMULATION RESULTS

The performance of the overall system has been assessed by means of the bit error rate (BER) after the LDPC decoder. The DVB-T2 parameters used in the simulations are: 64800 bits of length of the LDPC block, R = 2⁄3 of LDPC code rate, 16- QAM modulation, 2048 carriers as FFT size and 1/4 of guard interval. The simulations have been carried out over a Rayleigh channel (Typical Urban of six path, TU6), commonly used as the simulation environment for terrestrial digital television systems [22]. Perfect CSI has been considered at the receiver.

A. Candidate choice

When working with the ML metrics of LSD, i.e. the list L, the higher the number of candidates is, the more accurate the LE approximation is. Nevertheless, when considering the $\mathfrak{G}$ list of LFSD, the optimum value for N_{cand} will depend on the tree configuration vector n. Thus, the higher the value of n_T, the better the approximation is. In order to choose a suitable number of candidates for the detection algorithm, a battery of tests have been carried out. Fig. 3 depicts the bit error performance after the detection stage for a given SNR of 14.4 dB with different tree search configurations and N_{cand} values.

The effect of the ordering preprocessing stage is also depicted in this figure. The analyzed tree search configuration vectors $\boldsymbol{n}$ have been obtained by setting k = 1, 2, 3, which is equivalent to calculating $n_T = P^2$, $4P^2$, $9P^2$ Euclidean distances, respectively. On one hand, one can observe that the list ML approximation (5) converges for $N_{cand} > 30$. This involves that computing a very large number of candidates is not necessary in order to obtain a good $\boldsymbol{L_E}$ approximation of (4) for the proposed non-iterative scheme in Fig. 1. However, we should take into account that the exact computation of (4) provides a higher performance enhancement compared to applying the list ML approximation of (5) when iterative configurations are used [23].

On the other hand, a similar behavior for the fixed-complexity detector can be noticed, where the higher the value of k, the better the performance we obtain. Furthermore, it is noticeable that the ordering algorithm provides a performance enhancement such that the k = 2 LFSD approximates the BER values for the exhaustive MAP detector. Note that the BER degrades for a higher number of candidates with the tree search configuration k = 1.This is due to the fact that if we choose a large N_{cand} value from $n_T = P^2$ Euclidean distances, the probability of achieving the smallest or close to the smallest metrics is reduced. For k > 1, this effect is mitigated.

B. Performance comparison over DVB-T2 BICM

This section presents the performance assessment of the proposed list fixed-complexity soft detector over a SFBC DVB-T2 broadcasting scenario. The number of candidates considered for this study is N_{cand} = 50. Below graphs shows BER curves versus SNR for different configurations of the proposed algorithm in the detection of Golden and SS codes. For the Golden code, it is noteworthy that the ordering algorithm provides a gain of 0.4 and 0.25 dB compared to the nonordering case for $n_T = P^2$ and $n_T = 4P^2$, respectively. However, as previously stated, the ordering algorithm does not provide any performance gain with the SS code. In this case, the subchannel norms of the symbol pair (a, $\tilde{a}$) are completely equal, i.e., $\|h_b\|^2=\|h_{\tilde{b}}\|^2$ and $\|h_w\|^2=\|h_{\tilde{w}}\|^2$, being negligible the enhancement provided by the ordering procedure.

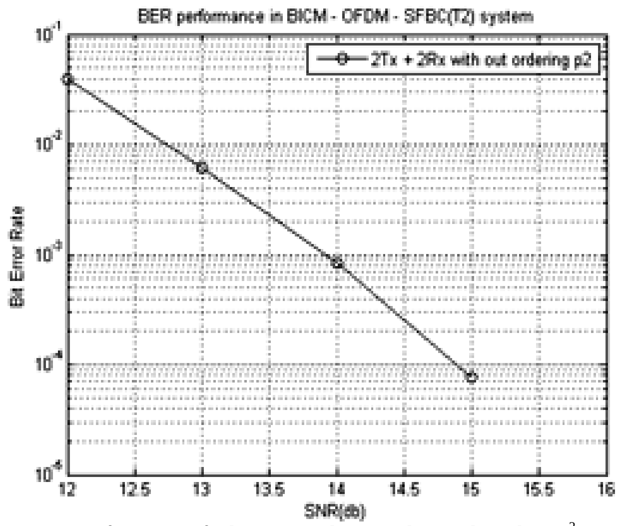

Fig 3. Performance of detector with out ordering algorithm (P^2)

The Fig 3. shows that by with out using the ordering algorithm for the equlidian distance (p^2), when BER (bit error rate) is 10^{-4} the signal to noise ratio (SNR) will be 14.9db, giving the normal performance for detecting the signals at different signal to noise ratio condition. Also it will be giving the average performance when detecting the signals in the different fading channel conditions like additive white gaussian noise and rayliegh channels.

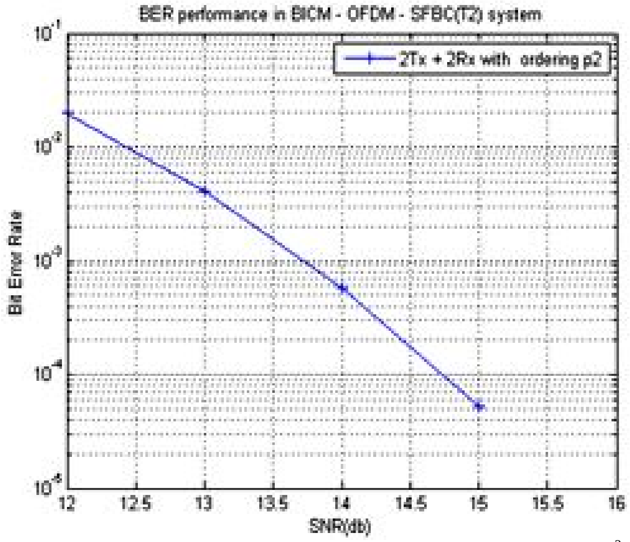

Fig 4. Performance of detector with ordering algorithm (P^2)

The Fig 4. shows that by using the ordering algorithm for the equlidian distance (p^2), when BER (bit error rate) is 10^{-4} the signal to noise ratio (SNR) will be 14.7db. This will give a better performance when compared with out ordering algorithm for detecting the signals at different signal to noise ratio condition. Also it will be giving the good performance when detecting the signals in the different fading channel conditions.

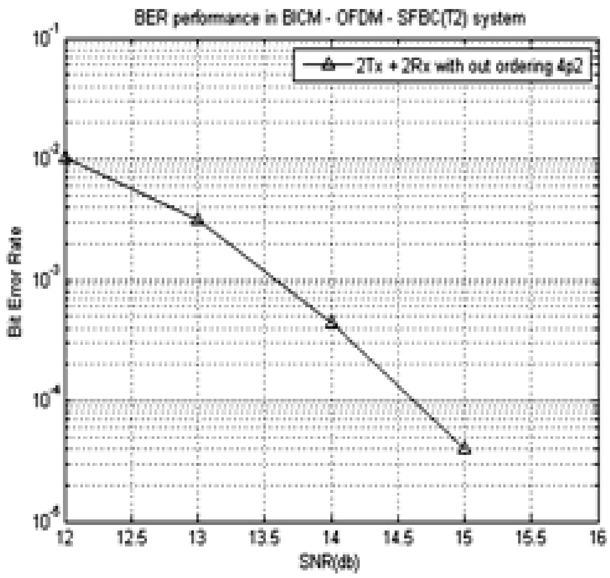

Fig 5. Performance of detector with out ordering algorithm ($4P^2$)

The Fig 5. shows that by with out using the ordering algorithm for the equlidian distance ($4p^2$), when BER (bit error rate) is 10^{-4} the signal to noise ratio (SNR) will be 14.6db. This will give a better performance when compared with out ordering algorithm for detecting the signals at different signal to noise ratio condition. Also in the different fading channel conditions like additive white gaussian noise and rayliegh channels, it will be better for detecting the signals with better quality.

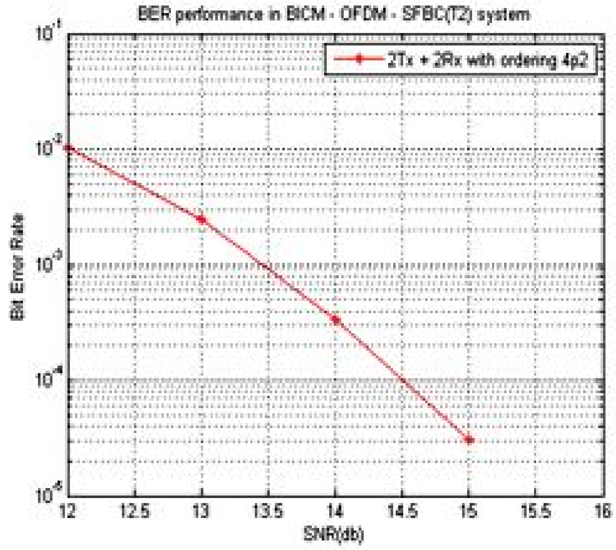

Fig 6. Performance of detector with ordering algorithm (4P2)

The Fig 6. shows that by using the ordering algorithm for the equlidian distance ($4p^2$), when BER (bit error rate) is 10^{-4} the signal to noise ratio (SNR) will be 14.5db. This gives the better performance when compared with out ordering algorithm, for detecting the signals at the different signal to noise ratio condition. Also it will be giving the better performance with reduced complexity and achieve full rate for detecting the terrestrial digital TV signals in the different fading channel conditions like additive white gaussian noise and rayliegh channels.

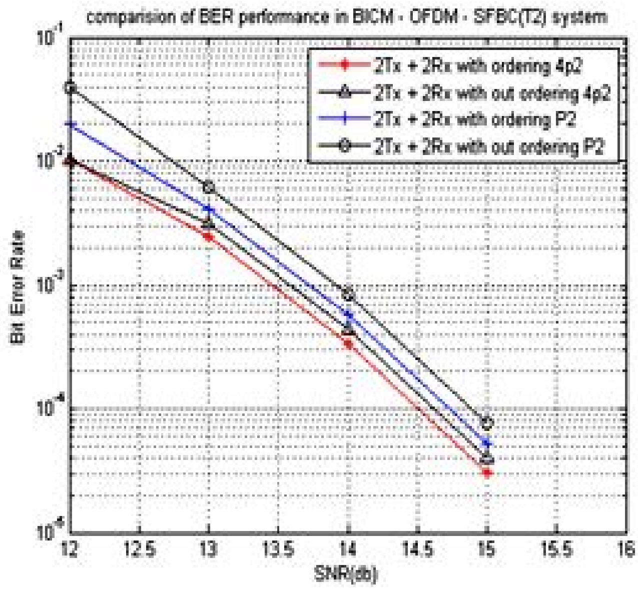

Fig 7. Performance comparision with and with out ordering algorithm

The Fig 7. shows that the performance Comparision based on different equlidian distances by using with and with out ordering algorithms.

V. CONCLUSION

Multi-antenna transmission using 2 × 2 FRFD codes, such as the Golden code or the SS code, increases the capacity allowing a higher data rate transmission with full diversity. This capacity increase involves a joint detection of a candidate list of four data symbol vectors in order to achieve soft information for the decoder. The complexity of this calculation can be reduced by means of different algorithms, being the most extended the LSD detector. However, the main drawback of this technique is its high and variable complexity, which can be upper-bounded by $\mathcal{O}(P^4)$. A list fixed-complexity detector with a novel ordering algorithm is proposed in this paper with the aim of approaching the performance of the LSD at a much lower complexity. Specifically, the complexity order can be reduced from $\mathcal{O}(P^4)$ to P^2 for the Golden code and from $\mathcal{O}(P^3)$ to P^2 for the SS code.

The analysis of the number of candidates shows that the list approximation does not need a high list size in order to converge to the exact soft information value. Provided simulation results show that a close-to-optimal detection can be achieved considering a reduced number of candidates (30 out of 65536 in 16-QAM). BER (bit error rate) simulation results show the close-to-optimal performance of the proposed low-complexity detector for both Golden and SS SFBC codes in a typical LDPC-based DVB-T2 broadcasting scenario. The performance is clearly improved when the proposed channel and candidate ordering algorithm is applied with Golden codes, though its effects are negligible for the SS code. In any case, the proposed detection algorithm can enable the realistic implementation and the inclusion of any FRFD SFBC code in any BICM-OFDM system such as the forth coming digital video broadcasting standards.

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Mobile Devices and Social Networking Security Solution

Aman Jain and Shilpa Mehta

Abstract: Nowadays almost everyone is using smart phones. They are becoming an essential tool in human being's everyday life. They are not only used for mere communication such as calling or sending text messages; however, they are also used in applications such as for accessing internet, receiving and sending emails and storing documents. As a result of this, not only phone numbers and addresses are stored in the mobile device but also financial information and business details which definitely should be kept private. And if the device is being stolen, each and every information is in the hands of the new owner. That's why; the biggest challenge is the security. When it is necessary to confirm the user identity on systems to perform a given operation, the term User Authentication is used. Traditionally, people prove their identity by providing passwords. The average person today has about 25 password protected accounts (according to Microsoft study), more passwords than they can reasonably be expected to remember. People compensate by using the same password for multiple accounts, and by choosing passwords that are easy to remember. But, unfortunately easy to remember means easy to guess. Other user select difficult passwords but then write them down where unauthorized eyes can find them. That's why, identity based on what you know (login and password) and what you have (ID cards) can be easily stolen. As we want trust (security), the notion what you are is a new opportunity to user authentication. Biometric Authentication is answer for that. Biometric is a characteristic of human being that distinguishes one person from another. For example, finger prints, retina, face recognition, etc. This can be used for identification or verification of identity.

Keywords: FAR, FRR, ROC, EER, FTE, FTC.

1. INTRODUCTION

Biometrics is a science of identifying a person based on unique physiological or behavioural characteristics. It is "who you are" type of authentication. Talking simply, biometric authentication works by comparing two sets of human features to figure out if they come from the same person. Our bodies have many features that are unique enough for such verification. Because biometrics can't be guessed, lost, shared, or stolen, it is the only way to reliably verify whether the person claiming to be you is really you.

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2. BIOMETRICAUTHENTICATION IN SOCIAL NETWORKS

A. The Advent of Social Networks

A social network service is a website designed to build communities based on common interest and activities. Typically users can create profiles with selected personal information and interests, search or view parts or all of the profiles of the other users, connect with old friends and make new ones. Social networking websites, such as Facebook, LinkedIn and Twitter have become a popular channel for communication.

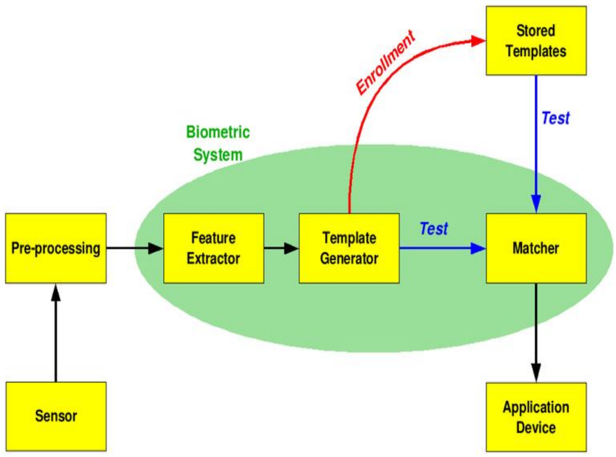


Fig. 1

B. Social Networking Today

Most major social networking sites today operate on the honour system. While many sites' policies require members to use their real name and age and to open only one account, there is no verification. Users can easily lie about their identity, with no traceability in case of a serious complaint. Many users give out too much personal information. Add to that tendency to use weak passwords, which can sometimes even be drawn from information in their profile, give away their login information and you have a huge receipt for abuse.

C. Biometrics for Social networks

The technology for such a system already exists, with a combination of human traits well suited to a social networking application. Biometric Identification of face, voice, iris, individually or in any combination, using standard computer webcam and microphone, such as the biometric technology developed by BioID , has been available for some time.

3. BIOMETRICAUTHENTICATION IN MOBILES

A. Security Aspects

There are two types of authentication: authentication at log on time and authentication at run time. The latter one is important because it can prevent unauthorized persons from taking device in operation and accessing confidential user information from the Personal Network. The false- accept rate (FAR) and the false reject rate (FRR) are used to quantify the biometric authentication performance.

B. Convenience Aspects

If the system force you to re- enter your biometric data, would you like? Never. Surely, you will get annoyed. The FRR is closely related to user convenience. A false reject will force user to re-enter the biometric data, which will cause annoyance. This obviously leads to the requirement of low FRR of biometric authentication system.

The biometric authentication should be transparent. Transparency should be considered as a requirement for the authentication at run time, because regularly requiring a user, who may be concentrating on a task, to present biometric data is neither convenient nor practical.

4. PERFORMANCE

The following are used as performance metrics for biometric systems

A. False Accept Rate or False Match Rate (FAR or FMR)

The probability that the system incorrectly matches the input pattern to a non-matching template in the database. It measures the percent of invalid inputs which are incorrectly accepted. In case of similarity scale, if the person is imposter in real, but the matching score is higher than the threshold, then he is treated as genuine that increase the FAR and hence performance also depends upon the selection of threshold value.

B. False Reject Rate or False Non-Match Rate (FRR or FNMR)

The probability that the system fails to detect a match between the input pattern and a matching template in the database. It measures the percent of valid inputs which are incorrectly rejected.

C. Receiver Operating Characteristic or Relative Operating Characteristic (ROC)

The ROC plot is a visual characterization of the trade-off between the FAR and the FRR. In general, the matching algorithm performs a decision based on a threshold which determines how close to a template the input needs to be for it to be considered a match. If the threshold is reduced, there will be fewer false non-matched but more false accepts. Correspondingly, a higher threshold will reduce the FAR but increase the FRR. A common variation is the Detection error trade-off (DET), which is obtained using normal deviate scales on both axes. This more linear graph illuminates the differences for higher performances (rarer errors).

D. Equal Error Rate or Crossover Error Rate (EER or CER)

The rate at which both accept and reject errors are equal. The value of the EER can be easily obtained from the ROC curve. The EER is a quick way to compare the accuracy of devices with different ROC curves. In general, the device with the lowest EER is most accurate

E. Failure To Enrol Rate (FTE or FER)

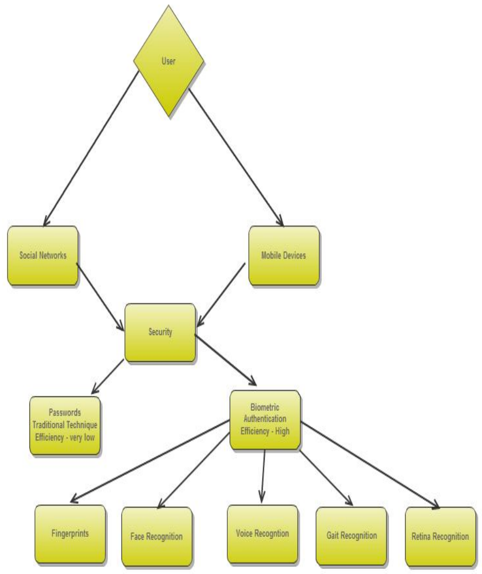
The rate at which attempts to create a template from an input is unsuccessful. This is most commonly caused by low quality inputs.

F. Failure To Capture Rate (FTC)

Within automatic systems, the probability that the system fails to detect a biometric input when presented correctly.

G. Template Capacity

The maximum number of sets of data which can be stored in the system.



5. FACE RECOGNITION AS BIOMETRIC AUTHENTICATION

Face recognition, among all biometrics, is a good choice. It is only biometrics that can be really transparent, especially for authentication at run time. Just by mounting a camera on the system, the face images of the user can be caught almost constantly. This is attractive also because the cost for the camera is low, and mobile phones and PDAs generally have a camera installed.

A. Face Pre-processing

It includes three steps, i.e. face detection, face registration, and face normalization.

For fast face detection Viola and Jones proposed a detection scheme which is very efficient by using simple rectangular binary features and the integral image, and has proved to be robust against varying background and foreground.

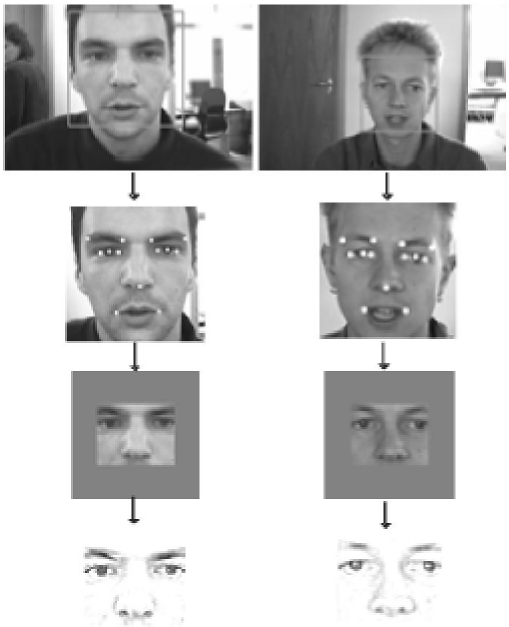


Fig. 2 Face feature extraction procedure (From top to down) face detection, facial feature detection, face registration and masking, high pass filtering

6. GAIT RECOGNITION AS BIOMETRIC AUTHENTICATION

The term *gait recognition* describes a biometric method which allows an automatic verification of the identity of a person by the way he walks. There are three main techniques in biometric gait recognition: Machine Vision Based, Floor Sensor Based and Wearable Sensor Based Gait Recognition.

A. Machine Vision Based Gait Recognition

In the machine vision based gate recognition, the system will typically consist of several digital or analog cameras with suitable optics for acquiring the gait data. Techniques such as background segmentation are used to extract features to identify a person. This technique is especially useful for surveillance scenarios.

B. Floor Sensor Based Gait Recognition

In the floor sensor based gate recognition, the sensors are placed on the floor which makes these methods suitable for controlling access to buildings. When people walk across the mat, they can be authenticated e.g. by the force to the ground which is measured by the mat.

C. Wearable Sensor Based Gait Recognition

The newest of three techniques is based on wearing the motion recording sensors on the body. It can be in different places: on the waist, in pockets, shoes etc.

We will mainly study this type of gait recognition.

The wearable sensors can be accelerometers (measuring acceleration), gyro sensors (measuring rotation and number of degrees per second of rotation), force sensors (measuring the force when walking) etc. Following Table gives overview of current wearable sensor based gait recognition studies from years 2004 to 2008.

Study	Sensor Location	EER	Number of Test Persons
Holien Gafurov et al. Gafurov et al. Gafurov et al. Huang et al. Ailisto el al.	Left leg (hip) Ankle Trousers pocket Hip Shoe Waist Waist	5.9%, 25.8% 8% 7.3% 15% 12% 7% 8%	60 40 55 100 40 15 10 35

Table I

Performance of Current Wearable Sensor- Based Gait Recognition Systems

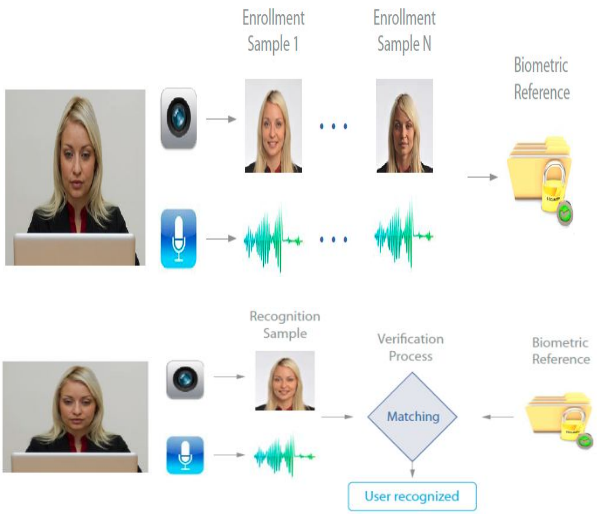
All studies except Morris and Huang et al. were using only accelerometers for collecting the gait data and reported recognition rates based on the verification criteria. Morris and Huang et al. used other types of sensors including force sensors, bend sensors, gyro sensors etc. in addition to the accelerometer sensor.

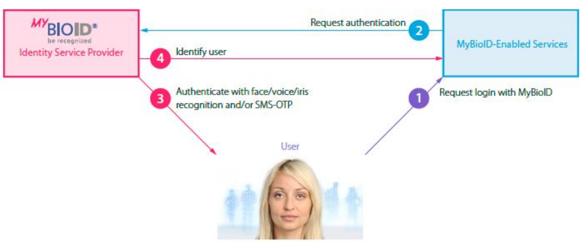
The accelerometer is focused for gait recognition for biometric authentication as it provides an unobtrusive authentication method for mobile devices which already contain accelerometers (like smart phones, PDAs etc.). That's why, it can be applied for continuous verification of the identity of the user without his intervention which is an advantage to other biometric authentication like fingerprint or face recognition.

The biometric gait recognition only works when user is walking. That's why this method has to be combined with another authentication method.

7. IMPLEMENTATION OF BIOMETRICAUTHENTICATION TODAY

A. Best example for Biometric Authentication in Social Networks is www.bioid.com





B. Various android applications are there for Biometric Authentication.

C. Unique Identification (UID) of Govt. of India uses Biometric Authentication.

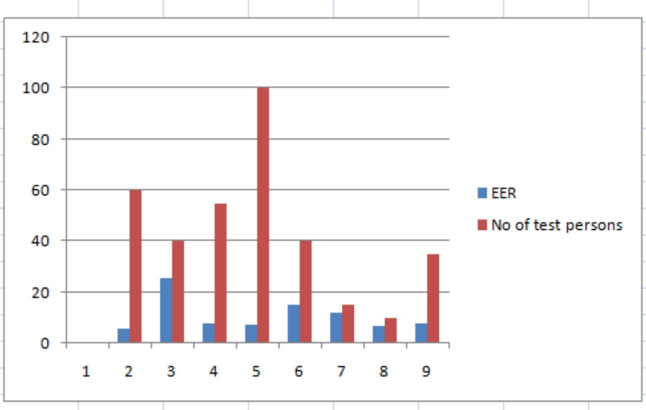
There used three biometric authentications i.e. face recognition, iris recognition and finger prints

D. It is also used in identification of culprits by matching fingerprints found in crime places with the police records or with that of suspects.

8. CONCLUSION

The security of the mobile devices is very important. My work aims at building up a secure and efficient connection between the system and the user, based on biometric authentication. A try has been made to create awareness for the biometric authentication. This paper mainly focuses for Face Recognition and Wearable Sensor Based Gait Recognition for Biometric Authentication.

9. RESULTS



10. FUTURE WORK

A. Improvement of Current Technologies

Both the face recognition and gait recognition have some advantages and disadvantages. Like, talking about the gait recognition, it main advantage is that it does not require user's intervention but its main disadvantage is that it only works when the user is walking so needs to be combined with other authentication method so not yet ready for practical use. Besides this, user does not walk always at constant speed and grounds can also be different.

So focus of future work is to create a gait recognition method which provides robust verification under different circumstances. These circumstances might be different walking conditions like walking speed or ground which will have an influence of the walk of a person and therefore might also influence the biometric recognition.

B. Implementation in Other applications

1. Facebook, LinkedIn and other social network sites should have option of logging by providing any biometric authentication. It will help the best to security concerned users.
2. Today smart phones require just pattern to be opened which is quite easy to guess. There must be employed face recognition where anyone try to operate the phone, his face is captured and automatically matched with the reference store. If both does not match, phone lock automatically.
3. Next step to Unique Identification (UID) Scheme Govt. of India can do, is that all the records must be provided to police, CID, CBI and every other investigating agency so that fingerprints found in crime places can be found easily. As sometimes, fingerprints found does not match with that in the police records, so this can be very useful.
4. High security building like Army Places etc. must be employed with Gait Recognition for Biometric Authentication.

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Recent Trends in Haptics Technology: A Simplified but Efficient Approach

Mr. Subhajeet Laga

***Abstract*— Modernization, although, has led into the development in terms of sophistication of human efforts in bringing out mass-production of more precisely manufactured goods; yet it in some ways it needs to be modified to be a smarter intelligent system. Intelligence in terms of decision making ability of machines, more specifically, robots. Modern day needs have broken the traditional belief of human interference just as an instructor to a machine or robot, and it is leaping towards an era demanding self motivated and corrective decision-maker automatons. In many situations, autonomous systems do provide effective solutions to menial or dangerous tasks. In many cases, it is desirable to design an automated system that can pick and relocate objects, engrave cravings, or replicate human arm gestures in order to facilitate designing in virtual and physical worlds. In our project we have implemented a similar but simplified system which will replicate human arm gestures to pick and displace objects.**

One may quote it as, "Bringing gestures of physical world into digital world, for solving tedious, repetitive and gigantic tasks; then reproducing the output back in terms of physical interpretation on objects".

Keywords—autonomous system, pick and place, human arm gestures, virtual and physical worlds.

I. Introduction

Today we are moving towards a world of automation and intelligent systems. The applications of electronics are, now only, limited to computers and communications. But they are influencing every field. In this day and age of computers, automation is becoming sensing, monitoring and storing the changes per millisecond involved in experiment with accuracy. Moreover, the repetitive tasks with same accuracy and sensitivity can be completed using automated instruments.[2],[4]

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There are many examples and live applications of these kinds of robots, a few of them are as:

1) The two-armed Canadian robot *Dexter* replaced a faulty circuit breaker outside the International Space Station late Monday, obviating the need to send astronauts out on an inherently dangerous spacewalk to perform the same work.[1]
2) The very cleverly named *Handroid* is a new robotic hand made by Japanese company ITK. It uses a system of tendon-like wires and their differential contraction moves every digit with precision. The hand is light and looks quite robust, in principal it appear similar to the robotic hand developed by DLR. Handroid is operated via a glove controller. The user wears the sensor-equipped glove and the robotic arm replicates his or her hand movement. Handroid can be used for remotely manipulating objects at dangerous conditions, as a prosthetic device or in various other applications.[3]

II. Problem Statement

Present day robotics is targeted towards development of autonomous robots or completely manually controlled robots. But none of these technologies deal with virtual and physical dimensions simultaneously. Hence our intention is to develop a robot which could be manually operated as well as can be grouped along virtual modeling, which in near future would be a boon to technical advancement in the field of robotics. [6],[4],[5]

Haptics is a technology which implements this approach. *Haptiks* stands for touch sense, in Greek. In this technology we implement a group of various sensors which altogether monitor various hand gestures through conductivity principle, i.e., we monitor touch sensing of the human arm. Then through the monitored data we implement another prosthetic arm to replicate the recorded gesture.[2] But this technology poses various problems in terms of implementation of sensors, their costs and their correctness. Hence, we have thought of and implemented a sensor less haptic system, which has same working principle and is much effective in terms of cost and reliability than the present day systems.

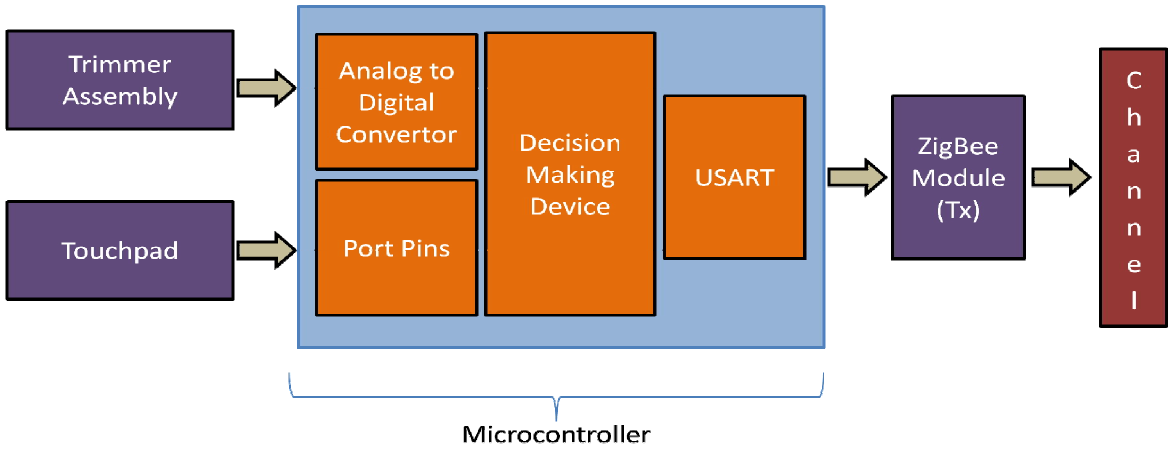

Figure 1: Haptic Section

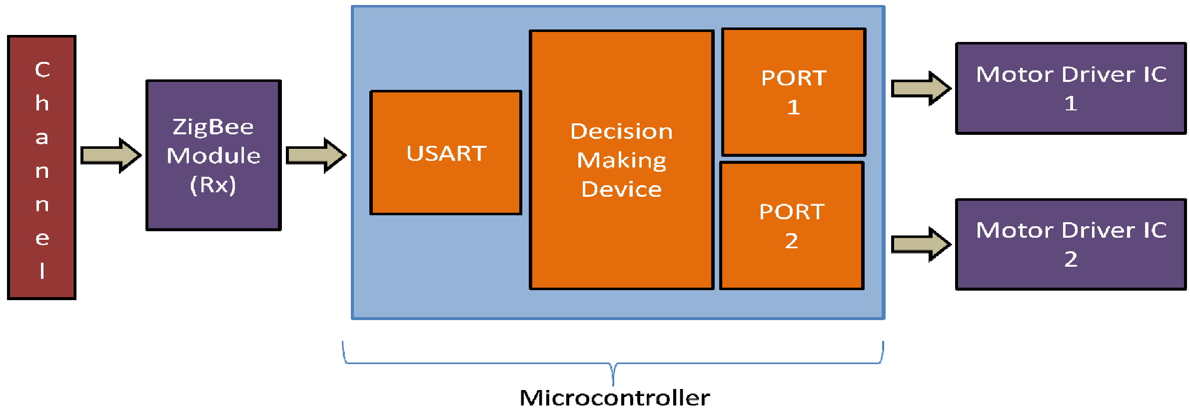

Figure 2: Replicator Section

Our interpretation for haptic sensors is through the utilization of potentiometers on various pre-recognized joints of the human arm. Basically, we have considered only five joints, initially for basic experimentation, and their respective motions and then made an assembly (glove) which consists of these potentiometers. The user wears this assembly/glove and then moves his hand. Due to which the knobs of the potentiometers is rotated. Hence the angular action of the joint is converted into change in resistance. Now this change in resistance is monitored using an analog to digital convertor (ADC). Thus we convert the angular action of the joint into digital value in terms of the ADC output.

III. Description of Block Diagram

A. *Trimmers*

Here in our system we are using five potentiometers, for sensing the various motions at joint coordinates. These trimmers are used to sense the rotational motions at the joint coordinates of hands. Also the trimmers convert the mechanical motions into electrical quantity, i.e., voltage.

B. *Touchpad*

The touchpad converts the touch sense into equivalent resistance. The touchpad is used to give locomotion command to the robot assembly. The output of the touchpad is applied to one of the ports of microcontroller.

C. *Analog to digital convertor*

The ADC is used to convert to the trimmer output, analog voltage, into digital code. This digital code is the interpretation of hand motion in terms of digital data. This code is necessary and sufficient data to be transmitted for replication of the hand gesture at the end effecter.

D. *Microcontroller*

The microcontroller is used take the ADC and Port data, and then it embeds them into a single code. This code is then serially transmitted through the UART module of the microcontroller. Thus microcontroller acts as data acquisition and coding block.

E. *ZigBee module*

The ZigBee module acts as a trans-receive device for serial communication. It receives the data and then sends an acknowledgment to the transmitter. One is configured as master and another as slave.

IV. SOFTWARE DESCRIPTION

This proposed hardware has two software routines, one at the transmitter end and another at the replicator end.

Algorithm of transmitter end routine:

a. Initialize peripherals like adc, lcd, serial port of P89V51RD2
b. Monitor the each channel of adc 0808 by polling method
c. Get the contents of adc output latch and transmit it through UART of microcontroller along with two sync characters
d. Convert the adc value the display in decimal format (for testing manually)
e. Repeat from step b onwards

Algorithm of replicator end routine:

a. Initialize peripherals like lcd, serial port of P89V51RD2
b. Monitor the serial port buffer
c. Check for two sync characters if check is correct proceed else wait
d. Save the all adc values and it in the display in decimal format (for testing manually)
e. Once all channel values are recorded go for next round of values
f. Compare old and new values find the difference
g. If difference is positive then rotate clockwise else anticlockwise (number of rotations is proportional to the difference)
h. If difference is zero then take no action
i. Repeat from step f onwards

V. ADVANTAGES

1. No image processing is required
2. Database management of stepping is not required
3. Real time execution is possible
4. Cost Effective system.

VI. APPLICATIONS

1. Can be used for remotely manipulating objects at dangerous conditions, as a prosthetic device or in various other applications
2. End-effectors can be equipped with three-dimensional force-sensors, providing the sense of touch. The surgeon seated at the workstation controls the robot using force feedback hand controllers.

VII. CONCLUSIONS

Thus, we have implemented the haptics concept without any complex sensors; thereby reducing hardware overheads and software complexity. Hence our experimentation can been viewed as a turning point in haptics in terms of simplistic approach with comparable results that to with reduced cost of implementation.

VIII. FUTURE SCOPE

To develop a virtual environment which could simulate the hand gesture in digital world and it would simultaneously monitor the replication on the physical object. Thus it would provide smarter and easier CAD Modeling.[4],[5]

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Stuck- at Fault Detection in Combinational Network Coefficients of the RMC with Fixed Polarity (Reed-Muller Coefficients)

Sharad Pratap Singh and Dr. B.B. Sagar

Abstract: **The paper presents a new method for computing all 2" canonical Reed-Muller forms(RMC forms) of a Boolean function and fixed polarity matrix method. The method constructs the coefficients directly and no matrix multiplication is needed. It is also usable for incompletely specified functions and for calculating a single RMC form. The method exhibits a high degree of parallelism. A fault we mean, in general any change in the value of an element with respect to its nominal value which can cause the frailest of the whole circuit. In the stuck-at model it is generally assume that a logic input or output is static or fixed to either logic '1'(stuck-at one) or logic '0' (stuck-at zero) and abbreviated as s-a-0 and s-a-1 respectively.**

Keywords: ***Fault detection, Read Muller Coefficients, Boolean function.***

I- INTRODUCTION

Digital systems are becoming increasingly complex and sophisticated development in LSI/VLSI technology which has allowed not only higher package density but also made ,it feasible to implement additional peripheral support devices that were earlier interconnected externally .in addition it permitted on-chip in corporation of novel advanced features such as self testing , supervisory and fail-safe etc. and hence simplified the task of design and maintenance of complex systems by reducing external interconnecting lines as also thought improved reliability. Many digital hardware building blocks such as Encoders/ Decoders, ROM / Demux, ROMs, PLAs and ASICs are frequently use to generate multiple output combinational function to implement digital systems intended for various application areas such as DSP, Automation , Control and computer systems etc. The increasing use digital systems in all aspects of social, economic and industrial applications have necessitated developed.

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II- METHOD FOR FAULT DETECTION:

Most commonly four methods are for used

1. Fault table method,

2. Boolean difference method

3. Path sensitization method and

4. D-algorithm

The advantage of this representation is the fact that the resulting circuit needs at most n inputs in contrast to up to 2n inputs in other cases. The second essential advantage is the fact that for each function represented in RMC form there exists circuit, which can be tested with maximum 3n + 4 tests, most of them independent of the realized function [2]. It is easy to see that for a Boolean function with n variables there exist 2" different RMC forms. Each of these forms can be characterized by 2" Boolean values a, indicating the presence or the absence of a given product term.

The aim is now to find the RMC form with the least number of a, = 1. Algorithms existing up to now build up a 2" x 2" matrix, called polarity-matrix [7], where every coefficient a, of each of the 2" polarities is given. This polarity-matrix is constructed using matrix multiplication [3], which means that these algorithms

Belong to the class with complexity AT2 = O (16") [I 3]

Method:

Definition: Let T n be a 2n x2n binary matrix. T n will recursively be defined as.

$$Tn = \begin{bmatrix} t^{n-1} & 0 \\ t^{n-1} & t^{n-1} \end{bmatrix} \quad \text{And To= [l]}$$

It becomes apparent that the same matrix can be obtained by the nth Kronecker-power [12] of T'.

Definition:

Consider a Boolean function f given as

[f] = [f′, f″]

Where, [f′] = [f0 ,……,f 2^{n-1}-1]

And, [f″] = [f 2^{n-1},….,f 2^{n-1}]

Then $2^n \times 2^n$ matrix B[f] is define as $B[F] = \begin{bmatrix} B[F'] & B[F''] \\ B[F''] & B[F'] \end{bmatrix}$ and B[fi] =fi

Let z^n be the nth kronecker power of $Z' = \begin{bmatrix} 1 & 1 \\ 0 & 1 \end{bmatrix}$

Then m[f] =B[f].2 z^n

$$\begin{bmatrix} P[0\,0\,1\,0] & P[1\,1\,1\,1] \\ P[1\,1\,0\,1] & P[1\,1\,1\,1] \end{bmatrix} =$$

$$\begin{bmatrix} P[00] & P[10] & P[11] & P[00] \\ P[10] & P[10] & P[11] & P[00] \\ P[11] & P[10] & P[11] & P[00] \\ P[01] & P[10] & P[11] & P[00] \end{bmatrix} =$$

$$\begin{bmatrix} 0 & 0 & 1 & 1 & 1 & 0 & 0 & 0 \\ 0 & 0 & 0 & 1 & 1 & 0 & 0 & 0 \\ 1 & 1 & 1 & 1 & 1 & 0 & 0 & 0 \\ 0 & 1 & 0 & 1 & 1 & 0 & 0 & 0 \\ 1 & 0 & 1 & 1 & 1 & 0 & 0 & 0 \\ 1 & 0 & 0 & 1 & 1 & 0 & 0 & 0 \\ 0 & 1 & 1 & 1 & 1 & 0 & 0 & 0 \\ 1 & 1 & 0 & 1 & 1 & 0 & 0 & 0 \end{bmatrix}$$

w1	3	4	3	3	3	5	2	3
a0	1	0	1	0	1	1	1	1
a1	1	1	1	1	0	0	0	0
a2	0	0	0	0	0	1	0	1
a3	0	0	0	0	1	1	1	1
a4	0	1	0	0	0	1	0	0
a5	1	1	0	0	1	1	0	0
a6	0	1	0	1	0	1	0	1
a7	1	1	1	1	1	1	1	1
w2’	4	5	3	3	4	7	4	5
w2	2	4	1	2	2	5	2	4
W=∑ wi	8	11	9	9	7	13	7	10

That computation of the coefficients of the RMC forms off for a given polarity is possible without constructing the whole matrix P. This can be advantageous in cases of lack of storage The 3-output
Polynomials with polarity (000) are as follows:
f0 = X1 ⊕XOX1⊕X2⊕ X0X1X2X3

f1 = X1 ⊕XOX1⊕X2⊕ X0X1⊕ XOX2X3

f2 = 1⊕X0⊕X0X2⊕ X0X2⊕ XOX1X3

P is an automorphism, because

P[f’] / P[f’’] = P[f’ / f’’] = P[f’’’]

P is defined as above, then P[f]=m[f]

P[f]=P[0 0 1 0 1 1 0 1

Inputs			Outputs		
X2	X1	X0	f1	F2	f0
0	0	0	1	0	0
0	0	1	0	0	0
0	1	0	1	1	1
0	1	1	0	0	0
1	0	0	1	0	1
1	0	1	1	1	1
1	1	0	1	1	0
1	1	1	0	0	0

TABLE-1 The number of Ex-OR gates for the example

POLARITY MATRIX FOR FUNCITONS F0, F1 & F2

P =	0	1	2	3	4	5	6	7
a0	0	0	1	0	1	1	0	0
a1	0	0	1	1	0	0	0	0
a2	1	0	1	0	1	1	1	1
a3	1	1	1	1	0	0	0	0
a4	1	1	1	0	1	1	1	0
a5	0	0	1	1	0	0	1	1
a6	0	1	0	1	0	1	0	1
a7	1	1	1	1	1	1	1	1
w0’	4	4	7	5	4	5	4	4

w0	3	3	5	4	2	3	3	3
a0	0	0	1	0	0	1	1	0
a1	0	0	1	1	1	1	1	1
a2	1	0	1	0	1	1	1	1
a3	1	1	1	1	0	0	0	0
a4	0	1	0	0	0	1	0	0
a5	1	1	0	1	1	1	0	0
a6	0	1	0	1	0	1	0	1
a7	1	1	1	1	1	1	1	1
W1'	4	5	5	4	4	7	4	4

Table.multipleoutputw'i denotes the number of nonzero coefficients of the ith polynomial, referred to as the weight of the ith polynomial, and wi denotes the number of exclusive-OR gates for realization of the ith polynomial.

Polarity	000 x2x1x0	001 x2x1x0	010 x2x1x0	011 x2x1x0	100 x2x1x0	101 x2x1x0	110 x2x1x0	111 x2x1x0
W_0	2	4	1	2	2	5	2	4
W_1	3	4	3	3	3	5	2	3
W_2	3	3	5	4	2	3	3	3
$W=\sum w_1$	8	11	9	9	7	13	7	10
$W^{(3)}$	0	2	1	2	0	3	0	2
$W^{(2)}$	2	1	2	1	1	1	1	1
Ns	2	5	4	5	1	7	1	5
N_{exor}	6	6	5	4	6	6	6	6

TABLE 2. The No. of NOT, AND & Ex-OR gates for the example

We see that the polarity (011) requires 4 Ex-Or gates for realization of the 3-output functions with regard to the common terms which is minimum than other polarity, hence polarity, hence polarity 3(011) is the optimum one. The number of input NOT gates, output NOT gates and EX-OR gates.

Polarity	000	001	010	011	100	101	110	111
N_{in}	0	1	1	2	1	2	2	3
N_{out}	1	0	3	0	2	3	2	0
N_{not}	1	1	4	2	3	5	4	3
N_{and}	3	4	3	4	3	4	3	4
N_{exor}	6	6	5	4	6	6	6	6

Table: 3-output function

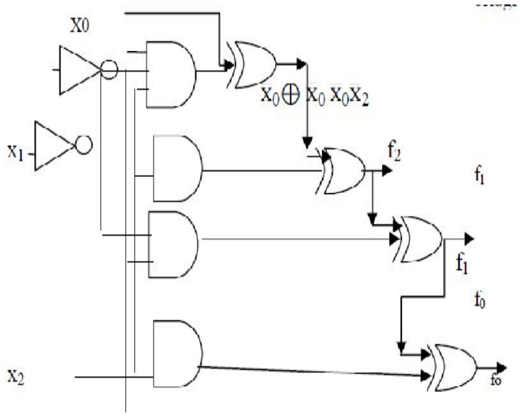

Fig : logical circuits for the example

III- PROPOSAL FOR FAULT DETECTION IN MULTIPLE OUTPUT CIRCUITS;

To detect the single/ multiple stuck-at- faults and bridging faults in multiple output systems, Its is proposed that above mentioned faults can be detected by verifying only one output function which contains all the input variables. If all the input variables do not appear in any of the output functions, then more than one function should be verified for fault detection. According to the proposal the faults in multiple output combinational circuit can be detected by considering the following cases.

Case1; When any one output function contains all input variables, then by testing only that function we can detect single/ multiple stuck-at faults and single bridging fault of n input circuit by verifying at most n, RM coefficients. The saving of test sets and time is highest in this case.

Case II;When in all output functions some input variables do not appear than we have to test more than one functions so that all input variables may be involved in fault detection. In this case larger test sets and computations have been claimed than in the case-1.

Case-III; When all the output functions are disjoint i.e., all functions have different input variables. Then all functions will be tested separately. It is the worst case in which largest test vectors and computations are required.

It has been reported that any R.M. network with k output and n input ($k \leq 2n$) requires at most 3n+5 test patterns to detect all single stuck-at faults and both AND and OR bridging faults which are detectable.

Concluding remarks:

a new method for the polarity matrix which is used for minimization of RMC forms of Boolean function is presented. It can be used for fully specified and incomplete specified functions.

In place of R. M. spectral coefficients techniques of fault detection, a more faster technique can be proposed to reduce further test sets and hardware overhead

P[f] matrix may be generated with further less no. of iterations, by modifying generation process of it.

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BER Analysis of BPSK based MIMO & MIMO-OFDM system

Pallavi Rahagude and Manoj Demde

Abstract: This paper focuses on BER analysis of BPSK based MIMO & MIMO-OFDM system. Nowadays, there is a requirement of higher data rate for wireless communication systems. In this paper we are going to compare MIMO & MIMO-OFDM system & how MIMO-OFDM is used to overcome high data rate problem is shown. MIMO is a multiple antenna technology. MIMO systems employ multiple antennas at both the transmitter & receiver to improve the range & performance of communication system. OFDM is a category of multicarrier modulation technique. In OFDM sub-carrier frequencies are orthogonal to each other i.e. they cannot interfere with each other (cross-talk between the sub-channels is eliminated). For transmission of signals over wireless channels OFDM is a very popular modulation technique. To reduce inter symbol interference (ISI) & to enhance system capacity OFDM for MIMO channels is considered.

MIMO & MIMO-OFDM module is carried out through Matlab simulation.

***Keywords:* intercarrier interference (ICI), intersymbol interference (ISI), multiple input multiple output (MIMO), orthogonal frequency division multiplexing (OFDM)**

I. INTRODUCTION

Increasingly, the driving force behind future growth in the telecommunication industries is seen to be broadband wireless access to the Internet and wireless data connectivity to mobile users. Multiple Input Multiple Output (MIMO) and Orthogonal Frequency Division Multiplexing (OFDM), which are new physical layer technologies capable of supporting the ever-increasing appetite for capacity and data rates, are the topic of this project. Multiple input multiple output orthogonal frequency division multiplexing has attracted a tremendous attention due to its large potential capacity and high-speed data rates. For the transmission of higher data rate more bandwidth is required. But due to the limitations of spectral bandwidth, it is often impractical or sometimes very expensive to increase bandwidth. For this purpose multiple transmit & multiple receive antennas are used for spectrally efficient transmission. The MIMO system can improve the capacity by a factor of the minimum number of transmit and receive antennas compared to single input single output (SISO) system with flat Rayleigh fading or narrowband channels.

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Because multiple data streams are transmitted in parallel from different antennas there is a linear increase in throughput with every pair of antennas added to the system. Unlike traditional ways of increasing throughput, MIMO systems does not increase bandwidth in order to increase throughput, it simply exploit the spatial dimension by increasing the number of unique spatial paths between the transmitter and receiver.

OFDM for MIMO channels is considered here which reduces intersymbol interference (ISI) & enhances system capacity. Intersymbol interference (ISI) is caused by multipath in band limited (frequency selective) time dispersive channels distorts the transmitted signal, causing bit errors at the receiver. The influence of ISI and the ICI generated by multipath environments can probably be removed totally in OFDM system. MIMO technology leverages multipath behavior by using multiple, “smart” transmitters and multiple receivers with an added “spatial” dimension to dramatically increase performance and range. Using MIMO multiple antennas can send and receive multiple spatial streams at the same time.

Orthogonal Frequency Division Multiplexing (OFDM) is a popular modulation scheme that is used in wireless LAN standards like 802.11a, HIPERLAN/2 and in the Digital Video Broadcasting standard (DVBT), metropolitan area network (MAN), digital audio broadcasting (DAB), it is also used in the ADSL standard.

OFDM and MIMO will serve as the physical layer of two key technologies for future mobile communication systems: UMTS LTE and WiMax. WiMax is a technology that is expected to deliver last mile wireless broadband access, while LTE is the 4G evolution of cellular systems.

II. PRINCIPLE OF MIMO

Various schemes that employ multiple antennas at the transmitter and receiver are being considered to improve the range and performance of communication systems. Today multiple-input multiple-output (MIMO) system is a most promising multiple antenna technology. MIMO system consists of multiple antennas at both the transmitter and the receiver as shown in Figure 1.

In conventional wireless communication, a single antenna is used at the source & another single antenna is used at the destination. In certain cases, this gives rise to problems with multipath effects. When an electromagnetic field (EM field) is met with obstructions such as hills, canyons, buildings & utility wires, the signals are scattered, & thus they take many paths to reach the destination. The late arrival of scattered portion of the signal causes problems such as fading, cut-off & intermittent reception. In digital communication systems such as wireless internet, it can cause a reduction in data speed & an increase in the number of error at the receiver. With the use of multiple antennas at the transmitter & receiver, along with the transmission of multiple signals (one for each antenna) at the source & the destination, eliminates the effect of multipath wave propagation .

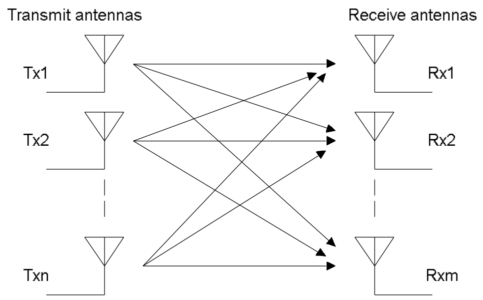

Fig. 1 General block diagram of MIMO

MIMO communication consists of multiple antennas at both the transmitter and receiver to exploit the spatial domain for spatial multiplexing and/or spatial diversity.

Spatial multiplexing is generally used to increase the capacity of a MIMO link by transmitting independent data streams in the same time slot and frequency band simultaneously from each transmit antenna, and differentiating multiple data streams at the receiver using channel information about each propagation path.

Spatial multiplexing techniques makes the receiver very complex, and therefore they are typically combined with Orthogonal frequency-division multiplexing (OFDM) or with Orthogonal Frequency Division Multiple Access (OFDMA) modulation, where the problems created by a multi-path channel are handled efficiently.

In short Multiple-input multiple-output (MIMO) wireless technology offers increased spectral efficiency through spatial multiplexing gain, and improved link reliability due to antenna diversity gain.

III. MIMO TECHNIQUES

As said before, the constant growth of data necessity by the mobile communication systems has motivated to develop new technique to give an answer to that demand. One of these techniques for increasing the amount of information that can be transmitted is MIMO. MIMO (multiple-input and multiple-output) techniques are part of the so called multi-antenna technologies. MIMO is currently use in WLAN (Wireless Local Area Networks) and is being considered as a candidate to be used for wider range wireless networks. Multiple antennas, both at the base station and at the mobile equipment, together with a sophisticated signal processing can improve drastically the performance of the wireless link, even under the worst case scenarios, without line of sight and fast moving mobile users. In a MIMO system diversity is a fundamental parameter to consider.

Lizhong Zheng y David N. C. Tse said that multiple antenna techniques can be used to increase the diversity or the degrees of freedom of the wireless communication systems. Their proposal was that both gains can be achieved at the same time, albeit there is a trade-off between them. The more independently faded channels we have the higher the diversity of the system will be. That effect is used for transmitting the same information through independent channels and average the received signal. The maximum diversity that is attainable in a system with {m, n} antennas (m transmitting antennas and n receiving antennas) is d = m . n. This will enable it to reduce the probability of error by a factor of 1=SNR compared to those of a normal system with just one antenna in transmission and one in reception, {m , n} = {1 , 1} where the probability of error will decrease by a factor of only 1=SNR. SNR stands for Signal to Noise Ratio and is the result of dividing the signal power by the noise power. The signals coming from different directions provide various degrees of freedom in communications. You can get the same effect even when the antennas are located nearby if there is dispersion. Essentially, if the channel response between pairs of transmit-receive antennas fades independently it can be said that there are multiple independent spatial channels. If we transmit independent information for each of these channels it can be increased the transmission rate. In a channel, {m,n}, with m transmitting antennas and n receiving antennas and rich scattering, there is min {m, n} degrees of freedom.

IV. PRINCIPLE OF MIMO-OFDM

Orthogonal Frequency Division Multiplexing (OFDM) is one of the most promising physical layer technologies for high data rate wireless communications due to its stoutness to frequency selective fading, low computational complexity and high spectral efficiency. OFDM can be used with a Multiple-Input Multiple-Output (MIMO) transceiver to increase the diversity gain and the system capacity by exploiting spatial domain. Because the OFDM system provides numerous parallel narrowband channels, MIMO-OFDM is considered a key technology in high-data rate systems means in future broadband wireless access such as Wi-Fi – 802.11n, Wi-MAX – 802.16e (a.k.a 802.16-2005), 3G / 4G.

Orthogonal Frequency Division Multiplexing (OFDM) and Multiple-Input Multiple-Output (MIMO) are cutting edge physical layer technologies slated to be employed in 4G wireless cellular standards such as 3GPP Long Term Evolution (LTE/LTE-A), Worldwide Interoperability for Microwave Access (Wi-MAX) and high speed WLAN standards. Such 4G cellular standards are visualized to support data rates in excess of 100 Mbps through OFDM, MIMO, dynamic carrier aggregation and thus enable a diverse plethora of applications in the wireless ecosystem such as broadcast/multicast video, HDTV on demand, high speed internet access and interactive gaming amongst others.

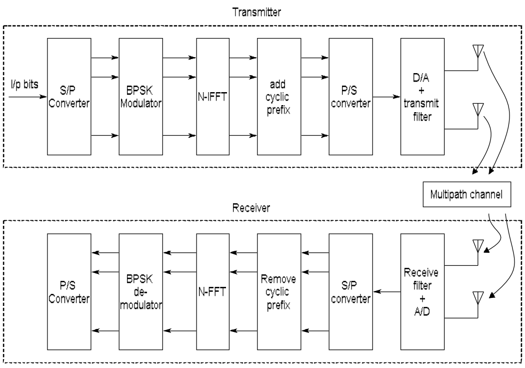

Fig.2 Block diagram of 2*2 MIMO-OFDM

The Orthogonal Frequency Division Multiplexing (OFDM) is very important transmission technique that belongs to the category of multicarrier modulation techniques. It solves the problems like ISI etc encountered by the conventional single carrier transmission techniques for high data rate transmission. In addition to this it provides higher bandwidth efficiency as compared to conventional multicarrier modulation techniques. Similar to conventional multicarrier modulation techniques, here the single high data rate stream is divided into several low data rate streams and modulated over different subcarriers. The only difference is the orthogonality of the subcarriers. The word orthogonal indicates that there exists a precise mathematical relationship (i.e. independence) between the frequencies of the subcarriers used in OFDM system. The subcarrier frequencies are selected in such a way that they are orthogonal to each other. Due to orthogonality property of subcarriers they can overlap each other in the frequency domain without interfering with each other and thus resulting in the higher spectrum utilization or greater spectrum efficiency, and also the data can be recovered on the receiver side without any Inter Channel Interference (ICI). That means it solves the problem of bandwidth requirement for data users.

V. RESULT

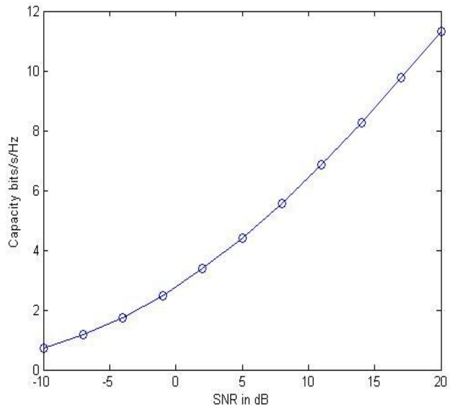

Fig.(a) SNR versus capacity bits/hz plot of 2*2 MIMO

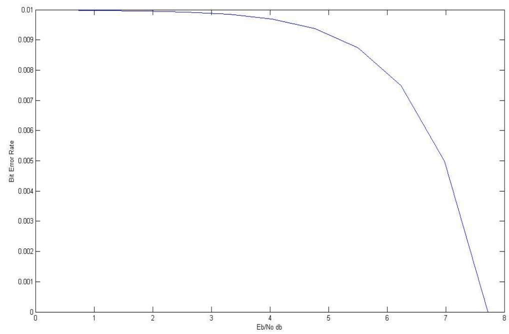

FIG.(B) BER VERSUS EB/NO DB PLOT OF 2*2 MIMO

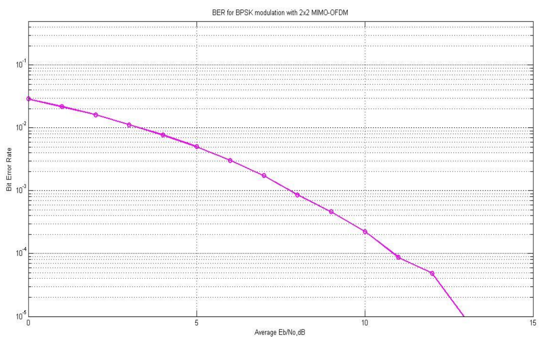

Fig.(c) BER versus Eb/No plot of 2*2 MIMO-OFDM

VI. CONCLUSIONS AND FUTURE WORK

Since radio resources are scarce and data rate requirements keep increasing, spectral efficiency is a rigorous requirement in present and future wireless communications systems. MIMO-OFDM has become a new star in the constellation of wireless and mobile communications. In addition to increasing spectral efficiency, MIMO can also be used to reduce transmitting power while keeping coverage areas constant. The use of MIMO technique in future transmission systems for broadcasting, multicasting and unicasting represents real business logic also for broadcasting corporations because of the possible reduction in transmission stations.

By using MIMO-OFDM technique BER rate will greatly reduce it is shown in the waveform which is necessary in new wireless applications.

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