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Abstract: Modern wireless communication applications are characterized by the need for advanced signal processing techniques such as Multiple-Input Multiple-Output (MIMO) technology for achieving high throughput and diversity and Orthogonal Frequency Division Multiplexing (OFDM) for achieving robustness to multipath fading. The implementation of such techniques at the transceiver level typically involves the design of algorithms with high processing complexity.

This paper introduces techniques that increase the throughput, reduces preprocessing delay and thereby increases the overall spectral efficiency of any wireless communication system. Generally preprocessing is done at the receiver in order to estimate the channel response. There are several preprocessing techniques such as Geometric mean decomposition (GMD), uniform channel decomposition (UCD), Singular Value Decomposition (SVD) Decomposition, and QR Decomposition. In an efficient preprocessed adaptive modulation and coding (AMC) design for MIMO–OFDM receiver systems, we propose PLU channel decomposition replacing the widely used QR decomposition. The PLU Decomposition algorithm is used to achieve a better matching of the processing rate of MIMO–OFDM receivers to the real-time processing deadlines imposed by the structure of the incoming data packets. The PLU decomposition algorithm is an attractive algorithm for MIMO–OFDM receiver because of its lower complexity, achieving the optimal operation point and it eliminates the need for buffering even for high number of antennas. Based on the Channel State Information acquired at the receiver, the AMC selector determines the modulation and coding pair for each sub carrier, which is then sent back to the transmitter. The modulation and encoding techniques for each subcarrier derived from the result of preprocessing is updated frame-by-frame to match the time-varying channel conditions, in order to take full advantage of the OFDM systems. The simulation results show that the proposed method is found to have lower computational complexity and better bit error rate performance than that of other conventional decomposition schemes.

Keywords: AMC, MIMO, OFDM, PLU, QR

I. INTRODUCTION

The performance of wireless communication can be drastically improved when using multiantenna transmission techniques. Specifically, multiantenna techniques can be used to increase the antenna gain and directionality (beamforming), to improve link robustness (space division coding), or to improve spectrum efficiency (space division multiplexing).

Orthogonal frequency division multiplexing (OFDM) has fascinated a great deal of attention due to its resilience to RF interference, high spectral efficiency [1] [2]. The combination of MIMO and OFDM has emerged as a promising choice for future high data rate wireless communications to achieve high capacity and high robustness without excessive complexity equalization, and thus MIMO-OFDM has been proposed for Wi-Fi, WiMax and 4G communication systems [3] [4].

MIMO technology for WiFi or WiMax offers the potential to take advantage of spatial diversity in an communication channel to increase the bandwidth without sacrificing larger portions of radio spectrum [5]. The general form of MIMO system consists of $n_t$ transmit and $n_r$ receiver antennas is shown in Fig.1.

Among the techniques of MIMO design, MIMO channel decomposition is one of the most important essentials since it governs how a mutually independent MIMO channel is decomposed into independent and scalarized sub channel gains. The optimization of processing at the receiver is of vital importance as it can reduce the symbol processing delays and related data buffering requirements and thus reduce complexity and cost of a MIMO-OFDM receiver. The various decomposition schemes are suggested with a compromise between performance and complexity [6] [7]. Among these
schemes, the most popular scheme is Singular value decomposition (SVD).

In the SVD, extremely widespread eigen values will suffer from two problems:
1. The large fading gain variation results in difficulties in coding and bit loading on each sub channel to attain the prescribed system bit error rate (BER).
2. The smallest eigen values limits the overall system performance.

The generalized Singular-value decomposition (GSVD) allows to use a single transformation for two different channels at one of the ends, but for each virtual parallel channel it yields a different gain for each user. Adapting SVD to this scenario is challenging since the decomposition requires multiplying by a channel–dependent matrix at the encoder which prevents from using this decomposition for MIMO-OFDM systems [8]. The

Geometric mean decomposition and Uniform channel decomposition have these problems:
1. Both the GMD and UCD have equal sub channel gains, bringing about low BER by using constant modulation and equal power allocation.
2. Another problem of GCD and UCD is that error propagation occurs at the decoder due to imperfect channel estimates [9].

The most widely used QR-based decomposition schemes fails.
1. It requires the individual streams to be simultaneously decodable at all the receiver implies that the rate per stream is governed by the smallest of the corresponding diagonal elements.
2. The preprocessing delay for the QR decomposition is large.
3. It needs data buffering for the higher number of antennas.

The existence of large preprocessing delays has a huge impact on the performance of the system. As a result of these difficulties, we propose the PLU decomposition algorithm, which reduce the symbol processing delay and also it eliminates the need for data buffering for the higher number of antennas. The PLU decomposition with the adaptive modulation scheme increases the data transmission rate. With the fixed modulation on channels with varying signal-to-noise ratio is that bit-error-rate probability performance is changing with the channel quality. Based on the channel state information, adaptive modulation and coding will select the modulation. Thus PLU decomposition method combined with adaptive modulation and coding effectively improves the spectral efficiency.

II. SYSTEM MODEL

We consider a spatial multiplexing MIMO-OFDM system with \( N_t \) transmit and \( N_r \) receive antennas, where \( N_t = N_r = N \). Although \( N_r \) can be larger than \( N_t \), we assume \( N_t = N_r \) for simple description. Then the received signal can be represented in discrete time as

\[
\mathbf{r}(k) = \mathbf{H}(k)\mathbf{s}(k) + \mathbf{n}(k) \quad (1)
\]

where \( \mathbf{r}(k) \in \mathbb{C}^{N_r} \) received signal vector for OFDM subcarrier \( k \) (where \( \mathbb{C} \) is the set of complex numbers), \( \mathbf{s}(k) \in \mathbb{C}^{N_T} \) is the vector of modulated signals departing from the transmitter at OFDM subcarrier \( k \) (where \( \mathbb{C} \) is the constellation corresponding to any modulation scheme such as Quadrature Amplitude Modulation), \( \mathbf{n}(k) \in \mathbb{C}^{N_r \times N_T} \) is the MIMO channel transfer function at subcarrier \( k \) that consists of the MIMO sub channels fading gains [10].

Receiver processing for MIMO-OFDM can be viewed as sequence of computational kernels (algorithms) connected in a pipelined fashion shown in Fig.2. This model consists of computation can be adapted to the processing architecture.

![Fig. 2. Typical MIMO-OFDM receiver](image)

The channel estimation takes place at the receiver by using the preprocessing algorithm. Here, the channel matrix $H$ can be decomposed as $H = LU$, where $L$ and $U$ are the lower and upper $N \times N$ triangular matrices, respectively [11].

Based on the channel state information acquired at the receiver the AMC selector determines the modulation and coding pair for each subcarrier, which is then sent back to the transmitter. The modulation and encoding techniques for each subcarrier derived from the result of preprocessing is updated frame-by-frame to match the time-varying channel conditions, in order to take full advantage of the OFDM systems.

III. PROPOSED PREPROCESSING ALGORITHM

The MIMO-OFDM receiver design uses a preprocessing algorithm in the channel estimation, in order to reduce the preprocessing delay.

A. PLU Decomposition Algorithm

PLU decomposition is a key function of the linear equation required for the MIMO demodulation. In addition, once LU decomposition is performed, the inversion of any nonsingular matrix $A$ can be easily obtained since $A^{-1} = (LU)^{-1} = U^{-1} L^{-1}$. Here, the inversion of upper (or lower) triangular matrix $U$ can be performed by directly solving $XU = I$, which leads to easy backward substitution [11]. The Fig 4 presents the pseudo-code for LU decomposition based on the pivoted LU LAPACK code [12]. In PLU, row permutations are not optimized because such a task is dependent on the target platform details.

The PLU pseudo-code has been arranged so that the algorithm operates on each column of the channel transfer matrix in succession (line 2: loop $j$).

Each iteration of $j$ (lines 3 to 24) can be performed after the corresponding $j$th column is estimated by the MIMO channel estimation process (line 2)[12]. Thus $PA = LU$, where $L$ is lower triangular and $U$ is upper triangular. It is also called as PLU factorization. We propose PLU channel decomposition replacing the widely used QR decomposition to accomplish a better matching of the processing rate of MIMO-OFDM receivers[13].

The pre-processing kernel is some form of matrix factorization of $H(k)$ such as QR- or LU-decomposition[14]. The PLU decomposition algorithm is an attractive algorithm for MIMO-OFDM receiver because of its lower complexity, achieving the optimal operation point and it eliminates the need for buffering even for high number of antennas. The AMC controller then updates the transmission mode at the transmitter shown in fig 3. Here the channel state information is obtained from the channel estimator at the receiver[15][16][17].

Receiver processing for MIMO-OFDM can be viewed as a sequence of computational kernels (algorithms) connected in a pipelined fashion. The channel estimation kernel involves estimation of the MIMO channel matrix per OFDM subcarrier[18].

In this way channel state is estimated and different modulation and coding scheme is used for different subcarriers. Then the adaptively modulated signals are then coded.

The cost of solving the linear equations is approximately $2/3 n^3$ floating point operations if the matrix $A$ has size $n$. This makes it twice as fast as algorithm based on the QR decomposition, which costs about $4/3 n^3$ floating point operations. For this reason, the LU decomposition is preferred[19][20].

Fig 3 : Adaptive MIMO-OFDM System
PLU Algorithm (column by column pipeline design)
Outputs: in place LU decomposition $A$, row perm. $I_{pivot}$

1. $I_{pivot} = [1 \cdots \min(N_R, N_T)]$ // initialize pivot indices

2. for $j=1, \cdots, \min(N_R, N_T)$
   /// get $j^{th}$ column from channel estimator
   
3. $a_j = h_j$ //Perform accumulated (trailing sub matrix) updates

4. for $k=1, 2, \cdots, j-1$
   exchange $(a_{kj}, a_{pivot(k)j})$

5. if $(k < \min(N_R, N_T))$ // undo row pivot affecting computations
   
6. $b = a_k$ // avoid permuting $A$

7. for $m=j, j-1, \cdots, k+1$
   exchange $(b_m, b_{pivot(m)})$

8. end // perform computations

9. for $m=k+1, k+2, \cdots, N_R$
   $a_{mj} = a_{mj} - b_m \cdot a_{kj}$

10. end

11. end // estimate row to be exchanged for current column

12. for $k=1, 2, \cdots, j$
   exchange $(a_{kj}, a_{pivot(k)j})$

13. end // compute elements of $a_j$

14. if $(j < N_R)$

15. for $m=j+1, j+2, \cdots, N_R$
   $a_{mj} = a_{mj} / a_{jj}$

16. end

17. end

18. end

Fig 4. Pseudo code for proposed PLU algorithm

IV. SIMULATION RESULTS
This algorithm is evaluated using the Network Simulator 2. An OFDM – MIMO systems with 8 transmit and receive antennas was simulated in NS2. The simulation was done with no preprocessing (using STBC codes), with QR preprocessing and with PLU preprocessing; with and without AMC.

The simulation parameters considered are

- $NUM\_DATA\_SUBCARRIER = 841$
- $MAX\_OFDM\_SYMBOLS = 100$
- $NUM\_SUBCARRIERS = 1024$
- $FFT\_size = 512$
- No of subcarrier used = 421
- No of sub channel = 15
- No of subcarrier per sub channel = 28
- No of data subcarrier per sub channel per symbol = 24

Table 2 compares the preprocessing delays between QR and PLU decomposition algorithms. It also shows improvement in throughput since processing delay is reduced and time is utilized for data transfer.

In Fig. 6 we can observe the throughput performance for a Non-Adaptive system corresponding to each modulation type in Table 2. For simulation 1, the combination parameters are: STBC 4x2, turbo code 1/3 and QPSK modulation; the maximum throughput is 160kbps with a starting point at SNR equal to -9dB. This means that a system that uses a fixed
modulation

Scheme without preprocessing can achieve a maximum throughput of 160 kbps within a relatively low SNR range; nonetheless, if we compare it with the other systems using different modulation schemes, 160 kbps results in the lowest maximum throughput among all the systems. The highest maximum throughput among all systems is achieved using the fixed 64 QAM. Nevertheless, there is a price to pay since the SNR necessary to guarantee such a maximum throughput is the highest among all the systems; about 5dB. Therefore, if a

<table>
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<th>Mod</th>
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<th>J</th>
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<th>Encoded data block size (bytes) = 48<em>n</em>M/B</th>
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Table 1: EESM Beta Values for Different modulation Technique

SNR necessary to guarantee such a maximum throughput is the highest among all the systems; about 5dB. Therefore, if a
In this paper, we proposed a system that combines AMC and MIMO schemes with PLU preprocessing method. The PLU preprocessing method improves the throughput of a MIMO–OFDM system by reducing the preprocessing delay, especially for large number of antennas. This, combined with adaptive modulation and coding, effectively improves the spectral efficiency.

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A Design & Comparative Analysis Of 320 Gb/s DWDM Optical Network With CSRZ, DRZ & MDRZ Modulation Formats

Gaurang H Patel, Prof. Rohit B Patel, Prof. Sweta J Patel

Abstract: This paper demonstrate design of 320 Gb/s Dense wavelength division multiplexing optical network with optimized modulation format. In this paper, system is simulated with 8-channels with data rate of 40 Gb/s. The system is simulated using different modulation formats and different dispersion compensation techniques like pre, post and symmetrical compensation. The optimized modulation format offer very high dispersion tolerance so due to this, it make possible to achieve longer distance communication. Among the different dispersion compensation techniques, symmetrical compensation with MDRZ modulation format shows best performance in terms of highest Q-Factor and minimum BER.

Keywords: BER, Dispersion Compensating Fiber, DWDM Network, Modulation Formats, Q-Factor.

I. INTRODUCTION

In today’s scenario demand for high bandwidth and high data rate is there because of growth in different technology like video on demand, use of internet, voice over IP, streaming video. Optical fiber fulfill this demand because it offer very large bandwidth so multiple channels can be transmitted through the common fiber using concept of wavelength division multiplexing technique. DWDM technique offers high spectral efficiency but performance is limited due to the problem of dispersion and fiber nonlinearities. So these problem must be minimized to achieve better performance. Dispersion compensation is achieved using different techniques like dispersion compensating fiber, fiber bragg grating, optical phase conjugation and electrical equalizer. Among all these technique, dispersion compensation fiber is proposed for compensation of dispersion. In conventional optical fiber communication, return-to-zero and non-return to zero modulation formats are used. But NRZ and RZ are not efficient for DWDM network. So different modulation formats like carrier suppressed return to zero(CSRZ) and modified duo-binary return to zero formats are proposed.

Bo-ning HU1, Wang Jing1, Wang Wei2 and Rui-mei Zhaol analyzed Fibres-optic dispersion and its effect on optical transmission system. In this paper, three schemes (pre, post and symmetrical dispersion compensation) of dispersion compensation with DCF are proposed. Symmetrical-compensation gives best result among all of these three [1].

M. I. Hayee and A. E. Willner describe the group velocity dispersion (GVD) and nonlinear effects, such as self- and cross-phase modulation (SPM/XPM) and four-wave mixing (FWM) in wavelength-division-multiplexed (WDM) systems at 10 Gb/s that degrade the performance of the system. In this paper, 10-Gb/s WDM systems that use pre-compensation, Post-compensation or dual-compensation of each channel to minimize dispersion and nonlinear effects is explained [2].

Anu Sheetal, AjayK.Sharma and R.S.Kaler describes the simulation analysis of 40 Gb/s long haul DWDM system with ultra high capacity has been carried out for carrier-suppressed return-to-zero (CSRZ), duo binary return-to-zero (DRZ) and modified duo binary return-to-zero (MDRZ) modulation formats. The DWDM system has been analyzed for the pre, post and symmetrical dispersion compensation schemes in order to find the optimum modulation format for a high bit rate optical transmission system [3].

Rajani, Raju Pal, Vishal Sharma investigate pre, post and symmetrical-dispersion compensation methods for 10/15Gb/s using different modulation formats like NRZ, RZ and RZ Super gaussian using standard and dispersion compensated fibers through computer simulations to optimize high data rate optical transmission. It is recommended to use symmetric- and post-DCF schemes for all the simulated optical pulses rather than using pre-DCF scheme at high transmission rate in dispersion compensated optical communication system in conjunction with laser line width of 100 MHz [4].

R.S. Kaler, Ajay K.Sharma and T.S. Kamala investigate pre-, post- and symmetrical-dispersion compensation methods for 10 Gb/s non-return to zero (NRZ) links using standard and dispersion compensated fibers through computer simulations to optimize high data rate optical transmission. The influence of EDFA power and increase in length of each type of fiber has been studied to evaluate the performance of optical communication systems [5].
II. DISPERSION COMPENSATION USING DISPERSION COMPENSATING FIBER.

In optical network, optical fiber offers very large bandwidth but it suffers from one problem of dispersion. Dispersion is nothing but it is broadening of the pulse in time domain due to the difference in the group velocity of different modes. It has two effects, 1) it reduces the energy contain in the pulse and 2) it results in spreading of pulse so it interfere with adjacent pulse so it creates inter symbol interference effect. There are mainly three types of dispersion. 1) Modal dispersion 2) Group velocity dispersion or chromatic dispersion and 3) Polarization mode dispersion. Modal dispersion is mainly occurred in multimode fiber because of the difference in group velocity of different modes. Chromatic dispersion is due to the material and waveguide property of the fiber. Polarization mode dispersion is due to the different polarization states of the mode travel with different group velocity. This dispersion problem degrades the system performance. So this dispersion effect should be minimized using different techniques to improve system performance. In this paper, to minimize dispersion effect, dispersion compensation fiber technique is proposed. Dispersion compensating fibers have negative dispersion of -80 to -90 ps/nm.km and used to compensate the positive dispersion of the single mode fiber. In optical WDM network, performance degradation is due to the chromatic dispersion, fiber nonlinearity, and accumulation of amplified spontaneous emission noise due to periodic amplification. Due to the nonlinear propagation of signal in optical fiber, system performance mainly decided by the power levels at the input of different types of fibers and also on the position of the DCF. There are basically three dispersion compensation schemes like pre, post and symmetrical compensation depending on the position of DCF in the system whether the DCF is placed before the SMF, after the SMF or symmetrically across the SMF. A DCF must have low insertion loss, low optical nonlinearity and also it must offer large negative dispersion coefficient to minimize the size of a DCF. By placing one DCF with negative dispersion after the standard fiber, the net dispersion should be zero.

\[ D_{\text{SMF}} \times L_{\text{SMF}} = -D_{\text{DCF}} \times L_{\text{DCF}} \]

Where \( D_{\text{SMF}} \) and \( L_{\text{SMF}} \) are the dispersion and length of single mode fiber and \( D_{\text{DCF}} \) and \( L_{\text{DCF}} \) are the dispersion and length of dispersion compensating fiber.

Compensation is done by three different methods depending on the position of the DCF:

(i) Pre-Compensation
(ii) Post Compensation
(iii) Symmetrical Compensation

Pre-Compensation: In this Compensation scheme, the dispersion compensating fiber of negative dispersion is placed before the standard fiber to compensate positive dispersion of the standard fiber.

Post-Compensation: In this Compensation scheme, the dispersion compensating fiber of negative dispersion is placed after the standard fiber to compensate positive dispersion of the standard fiber.

Symmetrical-Compensation: In this Compensation scheme, the dispersion compensating fiber of negative dispersion is placed before and after the standard fiber to compensate positive dispersion of the standard fiber.

III. DIFFERENT MODULATION FORMATS.

In this paper, three different modulation formats like carrier suppressed return to zero(CSRZ), duo binary return to zero(DRZ) and modified duo binary return to zero(MDRZ) are proposed. The different modulation formats and their simulation set-up is explained as below.

1) Carrier suppressed return to zero(CSRZ)

Fig. 1(a) shows the schematic diagram for the generation of the CSRZ modulation format. In this, the NRZ signal is given to MZM and then given to the phase modulator. The phase modulator is driven by a sine wave generator at the frequency half of the bit rate and phase shift of pi between any two adjacent bits is introduced. Because of this, the central peak at the carrier frequency is suppressed. It performs better in the presence of the combined effect of self phase modulation and chromatic dispersion. Fig. 1(b) shows the optical spectrum of CSRZ format.

![Fig 1: (a) schematic of CSRZ modulation format. (b) optical spectrum of CSRZ format.](image)

2) Duo-binary return to zero( DRZ)

Fig. 2(a) shows the schematic for the generation of the DRZ modulation format. In this first NRZ duo-binary signal
is generated by making the use of a duo-binary pre-coder, NRZ generator and a duo-binary pulse generator. The generator drives the first MZM and then connected with the second MZM. The second MZM is driven by a sine wave generator with the frequency of 40 GHz and phase of -90. DRZ has bandwidth half of the NRZ format. Fig. 2(b) shows the optical spectrum of DRZ format.

3) Modified duo binary return to zero (MDRZ)

Fig. 3(a) shows the schematic for the generation of the MDRZ modulation format. In this, first NRZ duo-binary signal is generated that drives the first MZM and then connecting this modulator with a second modulator that is driven by a sine wave generator with the frequency of 40 GHz and phase -95. Fig. 3(b) shows the optical spectrum of MDRZ signal.

IV. SIMULATION SETUP

In DWDM network, to achieve high capacity and high speed data transmission with higher accuracy, the dispersion and other non-linearity must be compensated. For this purpose, some dispersion compensation scheme must be used periodically in the link. There are several different methods that can be used to compensate for dispersion, including dispersion compensating fiber (DCF), fiber Bragg gratings, optical phase conjugation and electrical dispersion compensation. we have simulated one Dense Wavelength division multiplexing topology supporting four user and operated at data rate of 8×40 Gb/s=320 Gbps. In this topology, to compensate dispersion, dispersion compensation fiber (DCF) technology is used. We have simulated three DCF compensation scheme, pre-compensation, post compensation and symmetrical compensation scheme at 320 Gb/s with different modulation formats. Among these entire three schemes Symmetrical compensation scheme can greatly reduce the dispersion effects, this program is better than the pre compensation and post compensation program. The above techniques are compared in terms of Q-factor and Bit error rate. Fig.4 shows Block diagram of simulation setup of a 8 channel DWDM optical network at 320 GB/s. The simulation parameters and fiber parameters used in the system model are given in Table 1.
suppressed return to zero, duo binary return to zero and modified duo binary return to zero.

The transmission channel at 320 GB/s is designed by using the fiber parameters of DCF and SMF in such a way that the dispersion is compensated exactly. The gain of the erbium doped fiber amplifier (EDFA) placed after each fiber is such that it compensates the losses of the preceding fiber. The noise figure of the amplifiers is constant and set to 4 dB. The signal is then launched over 10 spans of standard single mode fiber (SMF) of 50 km each. DWDM system has been simulated for three different dispersion compensation schemes i.e. pre-compensation, post-compensation and symmetrical compensation. In pre-compensation scheme, as shown in Fig 4(a), to compensate the dispersion, DCF fiber of 10 km is used before the SMF fiber of 50km length. Also, two in-line-EDFA with gain of 5 and 10 dB have been used in the link. The post-compensation scheme has been shown in Fig 4(b) where DCF fiber of 10km is used after the SMF fiber of 50km length to compensate the dispersion. In symmetrical compensation scheme, as shown in Fig 4(c), two DCF fibers of 5 km are used before and after of the two SMF fibers of 25 km length each. Here four in-line-EDFA have been used in the link. In the receiver the signal is demultiplexed, detected by PIN detector, passed through the filter. The filtered electrical signal is given to the 3R Regenerator. 3R Regenerator output is connected directly to the BER analyser which is used as a visualize to generate graphs and results such as eye diagram, BER, Q value, eye opening etc. The parameters used in simulation are given in Table I.

Table I. Parameters used in simulation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Bit-Rate</td>
<td>320 Gbps</td>
</tr>
<tr>
<td>Length of SMF</td>
<td>50 Km</td>
</tr>
<tr>
<td>Length of DCF</td>
<td>10 Km</td>
</tr>
<tr>
<td>No. of spans</td>
<td>10</td>
</tr>
<tr>
<td>Dispersion coefficient of SMF</td>
<td>17 ps/nm/km</td>
</tr>
<tr>
<td>Dispersion coefficient of DCF</td>
<td>-85 ps/nm/km</td>
</tr>
<tr>
<td>Gain of Inline EDFA placed after DCF</td>
<td>5 db</td>
</tr>
<tr>
<td>Gain of Inline EDFA placed after SMF</td>
<td>10 db</td>
</tr>
<tr>
<td>Attenuation factor of SMF</td>
<td>0.2 db/km</td>
</tr>
<tr>
<td>Attenuation factor of DCF</td>
<td>0.5 db/km</td>
</tr>
</tbody>
</table>

V. SIMULATION RESULTS

We have simulated one DWDM network at 320 Gb/s with channel spacing of 100 GHz and the transmission distance is 500 km. we have made the comparative analysis of the system by employing different dispersion compensation schemes like pre-compensation, post-compensation and symmetric compensation and also using different modulation formats. The system performance is also analysed in terms of received maximum Q value and Minimum Bit error rate. The Eye diagram of the symmetrical compensation scheme with different modulation format is shown in figure below.
As shown in Fig. 4, Symmetrical compensation scheme with MDRZ modulation format gives highest Q-factor and minimum BER. The performance comparison of the different modulation formats with symmetrical compensation is shown in Table II.

We have also simulated pre compensation, post compensation and symmetric compensation with MDRZ modulation format and made the comparative analysis. The eye diagram of the above three dispersion compensation scheme with MDRZ modulation is shown in figure below.
modulation at transmitter side and symmetrical dispersion compensation scheme in the channel while designing the optical communication system.

REFERENCES


VI. CONCLUSION

In this paper, we have simulated one DWDM network at 320 Gb/s with channel spacing of 100 GHz up to distance of 500 km. We have made the comparative analysis of the different dispersion compensation schemes like pre, post and symmetrical with different modulation formats. From this comparative analysis, we have concluded that symmetrical dispersion compensation with MDRZ modulation format gives best performance than other compensation scheme and modulation formats. So it is preferable to use MDRZ
Defected Ground Structure Multiband Microstrip Patch Antenna using Complementary Split Ring Resonator

Jigar M. Patel, Shobhit K. Patel and Falgun N. Thakkar

Abstract: In this paper, Defected Ground Structure (DGS) microstrip patch antenna using Complementary Split Ring Resonator (CSRR) is designed and analysed. The aim to design such type antenna is to achieve multiband application which is the demand of current technology. Here Microstrip patch antenna with rectangle shape of patch with patch dimension 50×50 mm$^2$ is analysed. The proposed design is tuned with seven bands in the frequency range of 1–5 GHz depending on the geometric specification of antenna and the location of feed which can be used for multiband applications. Design results of VSWR, Return loss ($S_{11}$), Radiation Pattern are shown in this paper which is obtained by High Frequency Structure Simulator (HFSS).

Keywords- Microstrip, Multiband, DGS, CSRR.

I. Introduction

It is necessary to design multiband antennas to cover wide range frequency as per the requirement of mobile communication in current scenario. The design of such antenna has several advantages like light weight, small size, low cost fabrication and easy to integration with feed networks [1-10]. Microstrip patch antennas are mostly used in wireless device and other compact sizes with multiband applications. Microstrip patch antennas have narrow bandwidth and bandwidth enhancement is the practically application demand of today. In addition some applications of MPA required small size to meet the miniaturization in communication systems.

Defected Ground Structure is the one of the methods to use for this purpose. In this method the size of the antenna is reduced. DGS is realized by introducing a shape defected on a ground plane thus will disturb the shielded current distribution depending on the shape and dimension of the defect. The disturbance at the shielded current distribution will influence the input impedance and the current flow of the antenna. It can also control the excitation and electromagnetic waves propagating through the substrate layer. DGS is any defect etched in the ground plane of the microstrip can give rise to increasing the effective capacitance and inductance. DGS have the characteristics of stop band slow wave effect and high impedance.

DGS are basically used in microstrip antenna design for different applications such as antenna size reduction, cross polarization reduction, mutual coupling reduction in antenna arrays, harmonic suppression etc. DGS is widely used in microwave devices to make the system compact and effective. Therefore, in this paper we design a microstrip patch antenna with Defected Ground Structure for bluetooth to determine the effect of using DGS. Complementary Split Ring Resonator (CSRR) is used as DGS.

Complementary Split Ring Resonator is the dual of Spilt Ring Resonator (SRR), has been very popular resonator and widely used to synthesize metamaterial. Metamaterial exhibit qualitatively new electromagnetic response functions that cannot be found in the nature [11-37]. In 2004, Falcone et al introduced CSRR originally. The structure and its equivalent circuit model are shown in figure 1.

![Fig. 1. (a) Topology and (b) its equivalent circuit model of the CSRR [14]. Gray zone represents the metallization.](image)

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edges of the patch for radiation. The coupling between the CSRR and patch mainly comes from the capacitive coupling through the ring slot and the magnetic coupling through the split of the outer ring. By properly feeding the antenna, the inherent half-wavelength patch resonant mode can still be well excited. It is interesting to note that the interaction between the CSRR-inspired resonance and the patch resonance is very weak when they are orthogonally polarized. Under this condition circular polarization (CP) is attainable when they share the same operating frequency with a 90 phase delay in excitation. In addition, the interaction is strong when they are polarized in the same plane, which gives rise to two mixed modes.

It would be helpful to know the characters and design methodology for the CSRRs while designing the proposed CSRR-loaded patch antennas. The CSRR can be represented by an LC resonator tank as shown in Figure 1 when the loss is neglected. Its inherent resonance frequency is determined by

$$f_0 = \frac{1}{2\pi \sqrt{L_C C_r}}$$

where the capacitance \(C_r\) of the CSRR is approximately equal to that corresponding to a metallic disk surrounded and backed by the ground plane [14]. Here the inductance can be calculated based on a CPW structure with an equivalent perimeter of the CSRR, strip width \(w\), and slot width \(t\). The detailed properties for the CSRR is presented in [14], including the analytical calculation of the resonance frequency. However those equations are lengthy and calculation would become extremely difficult for irregular CSRR structures.

Recently a defected ground structure (DGS) has been introduced, DGS is realized by etching a simple shape in the ground plane, depending on the shape and dimensions of the defect, the shielded current distribution in the ground plane is disturbed, resulting a controlled excitation and propagation of the electromagnetic waves through the substrate layer. The shape of the defect may be changed from the simple shape to the complicated shape for the better performance. Defected Ground structures (DGS) have two main characteristics slow wave propagation in Pass band & Band Stop Characteristics in microwave circuits [17]. In order to explain the cutoff and the attenuation pole characteristic of the proposed DGS section simultaneously, the equivalent circuit should exhibit performances of low-pass and bandstop filter at the same time [18]. Generally it is accepted that the microstrip line should have the impedance around 100–130 ohms. By using the defected ground structure in the ground plane the effective inductance will increase and at the same time the capacitance will be decreased and finally the impedance of the transmission line increases and becomes more than 200 ohms. This high impedance of the DGS is used in the interconnects used in the digital systems [18].

Based on the idea of photonic band-gap (PBG) structure, defected ground structure (DGS) was firstly proposed by Park et al. in 1999, and has found its application in the design of planar circuits and low pass filters [19]. DGS is realized by etching a defective pattern in the ground plane [20], which disturbs the shield current distribution in the ground plane. This disturbance can change the characteristics of a transmission line such as equivalent capacitance and inductance to obtain the slow-wave effect and band-stop property.

II. Design and Modelling

This section, we will introduce the design of our antenna. First the conventional patch length and width is designed. After designing the patch, we have taken CSRR as DGS in the ground. Basic length and width is designed with the use of following equations.

$$w = \frac{c}{2 f_0 \sqrt{\varepsilon_r + 1}}$$

(2)

The width of the patch can be designed using the equation (2), here \(f_0\) is the center frequency, \(\varepsilon_r\) is relative permittivity and \(c\) is speed of light.

$$L_{eff} = \frac{c}{2 f_0 \sqrt{\varepsilon_{off}}}$$

(3)

$$\varepsilon_{off} = \frac{\varepsilon_r + 1}{2} + \frac{\varepsilon_r - 1}{2} \left( \frac{1}{\sqrt{1 + 12t}} \right)$$

(4)

$$\Delta L = 0.412t \left( \frac{W}{t} + 0.264 \right)$$

(5)

$$L = L_{eff} - 2\Delta L$$

(6)

Length of the patch can be designed by using the equations (2-6). Here \(t\) is the thickness of substrate. Using these equations we have designed length and width of conventional patch here.

Here we designed square patch so length and width are same and it is 50 mm, so a square patch is 50×50 mm² over here which is shown in Figure 1. We have taken out CSRR as DGN in the ground and to improve the results as shown in figure. The CSRR taken out have dimension as shown in figure 4(a). The top view and side view of the design is shown in Figure 4(a) and 4(b) respectively.
Table 1 shows details about the material. Patch is of copper material. Substrate is of FR4 epoxy material with $\varepsilon=4.4$. The base material is also of copper.

### III. Simulation results and discussions

For simulation we used HFSS 11 of Ansoft, which is very good simulator for RF antennas. After simulating the design the result we got is as follows. Figure 5 shows the Return Loss ($S_{11}$) plot of the design and Table 2 shows values of Return Loss ($S_{11}$) in dB for different bands with their frequency. The minimum return loss which we are getting for this design is -31.25 dB for the second band centred around 1.9423 GHz.

![Fig. 4 (a) Top View of Proposed Antenna](image)

![Fig. 4 (b) Side View of Proposed Antenna](image)

**Table 2: Return Loss ($S_{11}$) values**

<table>
<thead>
<tr>
<th>Band</th>
<th>Frequency in GHz</th>
<th>Minimum Return Loss ($S_{11}$) in dB (Negative Values)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1&lt;sup&gt;st&lt;/sup&gt;</td>
<td>1.3538</td>
<td>12.8846</td>
</tr>
<tr>
<td>2&lt;sup&gt;nd&lt;/sup&gt;</td>
<td>1.9423</td>
<td>31.25</td>
</tr>
<tr>
<td>3&lt;sup&gt;rd&lt;/sup&gt;</td>
<td>2.6038</td>
<td>15.3846</td>
</tr>
<tr>
<td>4&lt;sup&gt;th&lt;/sup&gt;</td>
<td>3.0538</td>
<td>21.25</td>
</tr>
<tr>
<td>5&lt;sup&gt;th&lt;/sup&gt;</td>
<td>3.8769</td>
<td>12.2115</td>
</tr>
<tr>
<td>6&lt;sup&gt;th&lt;/sup&gt;</td>
<td>4.2962</td>
<td>12.9808</td>
</tr>
<tr>
<td>7&lt;sup&gt;th&lt;/sup&gt;</td>
<td>4.8654</td>
<td>11.1538</td>
</tr>
</tbody>
</table>

Figure 6 shows the voltage standing wave ratio (VSWR) plot of the design and Table 3 shows values of...

![Fig. 5 Return Loss ($S_{11}$) parameter of the antenna](image)

![Fig. 6 VSWR of Antenna](image)
VSWR for different band with frequencies. For the entire band VSWR is less than 2 and lowest VSWR for the design is 1.2363 for the second band centered around 1.9423 GHz.

<table>
<thead>
<tr>
<th>Band</th>
<th>Frequency in GHz</th>
<th>VSWR</th>
</tr>
</thead>
<tbody>
<tr>
<td>1st</td>
<td>1.3538</td>
<td>1.6484</td>
</tr>
<tr>
<td>2nd</td>
<td>1.9423</td>
<td>1.2363</td>
</tr>
<tr>
<td>3rd</td>
<td>2.6038</td>
<td>1.5110</td>
</tr>
<tr>
<td>4th</td>
<td>3.0538</td>
<td>1.3736</td>
</tr>
<tr>
<td>5th</td>
<td>3.8769</td>
<td>1.6484</td>
</tr>
<tr>
<td>6th</td>
<td>4.2962</td>
<td>1.5110</td>
</tr>
<tr>
<td>7th</td>
<td>4.8654</td>
<td>1.7857</td>
</tr>
</tbody>
</table>

Figure 7 shows the radiation pattern of the design which shows the total gain of frequency range 1 - 5 GHz.

The Total gain of antenna in dB is shown in figure 8 which is in 3D View. The Total gain of antenna is -5.8802 dB as shown in 3D Polar Plot. Figure 9 shows the total directivity of the antenna which is -3.2921 dB as shown in 3D Polar Plot.

Figure 10 and 11 shows the magnitude plot of E - Field and H – Field respectively of the Antenna. As shown in figure the magnitude value of E – Field is 2.2318e+004 and the magnitude value of H – Field is 7.0881e+001.
IV. Conclusion

Microstrip antennas have become a rapidly growing area of research. Their potential applications are limitless, because of their light weight, compact size, and ease of manufacturing. Here DGS microstrip patch is designed for multiband applications. The modelling and iterative simulations are carried out at centred frequency of 2.5 GHz. The result indicates the three bands so the antenna can be used for L Band and S Band Applications. Further design can be modified to have a multiband for other applications in C Band, X Band and other bands. The results are in very good agreement with the industry and standard published antenna-requirements with respect to ease of fabrication, compactness and volume miniaturization compared to other antennas so far designed for similar applications. Instead of natural material, the artificial material, meta material can be used to improve the result and also using multilayer substrate can be used to improve the result.

References


Abstract- Radio Frequency Identification (RFID) technology has applications from retail sector to health sector. This paper presents a low frequency RFID based Object Identification System (RFASSIST) that has been produced to help blind people to identify various objects. RFID appears as an effective solution to provide object identification. The paper also gives an overview of the RFID technology. RFASSIST uses an 8-bit PIC Microcontroller to interface RFID Reader Module. RFID reader tracks an object carrying passive RFID tag in an indoor environment. Microcontroller processes signals received from RFID reader. To provide assistance to the blind, the framework combines RFID based object identification with audio messages. It also displays the object’s name on LCD. This project stores the date and time of identification of object on PC using RS232 serial communication. The data was successfully stored in computer’s memory which can be later used as a database in certain applications. During the course of the project indoor experiments have been conducted. This system successfully identified various objects for application domains such as home care, library management, toll gate etc. for blind.

Keywords- Audio messages, Low frequency, Microcontroller, Object Identification, Radio Frequency Identification (RFID) technology, RS232 serial communication.

1. INTRODUCTION

Sense of Sight is one of the basic senses of a normal human being. It renders life quite difficult for a blind person. Visually impaired people have difficulty in navigating in unknown campuses. As well, they must have every detail about indoor environment. Large obstacles can cause them injuries, so they should be kept aside. They also have difficulty in locating various objects. For them object identification and physical movement is one of the biggest challenges [5], [8], [11], [16], [21]. Many people with serious visual impairments can identify objects and navigate independently, using a wide range of tools and techniques. For providing assistance to differently abled person several devices such as One click method-arm approach [2], Wireless sensor network [3], Automatic speech recognition [15], Finger Braille [23]. RFID [1,4,8,9,10,11,12,13,16,18,20,21,22,24] have been proposed. Amongst these RFID appears to be an effective solution for the visually impaired people.

Radio Frequency Identification (RFID) is a method of storing and remotely retrieving data via a radio frequency transmission by using devices called RFID readers (receivers) and RFID tags (transponders). RFID system mainly operates in four ranges of frequency spectrum: low, high, ultrahigh and microwave. Low frequency RFID systems operate at 125 KHz and have a read range of less than 0.5m. High frequency systems operate at 13.56 MHz and provide a read range of approximately 1 m. Similarly, Ultrahigh and Microwave frequency systems operate at 860 MHz and 2.4 GHz and provide read range of 3m and 1m, respectively. RFID tags are inexpensive, bar-code sized stickers that contain an antenna and a microchip that can be sensed wirelessly by an RFID reader. Information on RFID tags usually contains at least a unique id. When RFID reader transmits a signal to the tag, tag communicates its identity to the reader. The information can be read and sometimes written at distances of up to 30 feet, depending on the system. Tags are of two types: Active and Passive. Active tags possess batteries that supply power for their communication circuitry [1]. They have long wireless range and can achieve high data rates. In contrast, passive tags don’t have interior power supply and they rely on interrogator for power. Thus, they are not limited by battery life. There are numerous application specific and vendor specific tag types. Common formats include disk or coin tag, smart label, contact-less Smart Card. There are many other available formats like plastic housed, coil mounted tags etc. A detailed review of RFID Technology and its design and implementation is given by the authors et. al. [12], [24]. RFID has become widespread in retail and shipping where it is used as an alternative to bar codes to detect and identify products and shipments. In recent years, RFID based identification have become popular in applications such as security systems, library management, vehicle security, electronics barcode labels and toll systems [9].

This work implements a low frequency RFID based object identification system for the blind people (see Fig. 1). In indoor environment, blind people can identify several objects using RFID reader which identifies objects carrying passive RFID tags. This information is relayed to the microcontroller, which then plays a pre recorded message when the object is identified. This information can also be stored on PC using RS232 communication interface. Hence, this system can be used in several applications like library management. It also shows opening and closing of gates upon tag identification. Thus, this system can also be used in security and toll systems. Many experiments have been successfully conducted to demonstrate such applications. Results have also been compared with the existing systems and plotted. Installation is easy, development cost is low, and efficiency is good below 0.5m. The other advantage is that it is applicable in indoor environment. This paper is
organized as follows: Section 2 gives the review of various RFID applications; Section 3 describes the system hardware and software architecture and its functionalities; Section 4 describes various experiments conducted and derives results. Finally Section 5 concludes the paper and proposes future work.

2. RELATED WORK

For related researches on RFID [1], [4], [5], [8], [9], [10], [11], [16], [18] and [22] are the prominent examples. RFID systems have become common for identifying and tracking objects as discussed before. Such systems have been proposed in [4], [5], [8], [11], [16]. Elif et al. [4] proposed RFID based Moving object tracking system. This paper presents a study on localization and tracking of an object carrying an active RFID tag. The Study incorporates processing of the signals received from transmitters, in a way to locate and track the coordinate in an indoor environment. Here, Bayes Decision Rule and Kalman Filter approach were introduced separately to improve the performance of location estimation and tracking in presence of obstacles. They provided a read range of 2m and 5m respectively. However, Received Signal Strength (RSS) was influenced by scattering and reflection of radio waves. It resulted in low accuracy. Ersin et al. [5] developed An RFID application: Path Finder, which was very accurate to allow the user group to reach the target within a non familiar or totally unknown campus area without taking any help from outside. But the Path Finder could not achieve expected results in case of more than one target. Also tags were not read if they were far away than 35cm. Read range was improved drastically in the system developed by Mohtsin [11]. He developed An RFID Based Navigation and Object Recognition Assistant for Visually Impaired People. This system was proposed for the assistance of blind people. The system incorporated a mobile RFID reader module with object management system with three main functionalities: to access and manage targeted an integrated Zigbee transceiver for transmitting the tag’s information. Utensils and other objects in the house or building carry the tags and transmit the data wirelessly to the server embedded. An audio file, recorded for and unique to each object, resides on the server. The reader reads PC which in turn scans for the particular ID in the database and plays the corresponding audio file. This system provides excellent voice quality within 100m. But it requires refinement and use of accurate reader modules to handle interference problems. The system is expensive because of use of Zigbee transceivers. Cost was greatly reduced in the system developed by Sakmongkon et al. [16]. He developed Blind Navigation System for Indoor Environments using UHF RFID systems. This system uses a passive communication circuitry and is capable of locating tags up to 10-15 m. The cost in this system was reduced as the connection to the server was only when navigation starts. In this paper rechargeable batteries were used which greatly removed the problem of limited battery life. But some communication delay was observed. The transponder cost was greatly reduced by reusing obsolete transponders available from livestock identification by Lorenzo Faggion [8] which are also fully operable in different adverse environments. The paper proposed by him presented low frequency RFID based Mobility Network for Blind People. The use of Smart phones resulted in compatibility and portability problems associated with it. A visual aid system (VAS) for blind was also developed based on RFID and fast symbol recognition which achieved good performance in embedded devices. VAS perceives all blind symbols in current scene and reports location information to blind by text-to-speech technology. This helps to achieve real time effects.

Other RFID applications also exist which result in low cost and improved battery life. Alanson et al. [1] presented the Wireless Identification and Sensing Platform (WISP). WISP is a programmable battery free sensing and computational platform designed to explore sensor-enhanced radio frequency identification (RFID) applications. It exclusively operates from harvested RF energy and uses a 16-bit ultralow-power microcontroller to perform sensing and computation. The microcontroller encodes compliant ID and dynamically computes the required 16-bit cyclical redundancy checking (CRC). But in this, Power consumption is high and there is loss in signal strength over transmission. Martin et al. [9] developed low cost wireless sensor system and presented its application in dental retainers. Use of wireless links reduced high risk of inflammatory disease but battery capacity and power consumption limited its performance. Other applications include object management system and mobile telecommunication services. Meenakshi et al. [10] focused on the applications of RFID technology in mobile telecommunication services. Here, mobile phone can be both tag and reader at the same time but economic and standardization issues need to be handled. Taketoshi et al. [18] developed indoor...
objects, real time monitor locations and search particular information from database. In this, floor embedded pressure sensors reduces interference from obstacles and computation time but it makes use of complex algorithms.

From the above discussion it is concluded that RFID is a powerful tool for providing assistance to the blind. There is a possibility of designing a cost effective system which has an improved performance in most of the respects and will also work optimally in many different applications. RFID based systems appeared to be the best promising solution and easy to use.

In the nutshell, RFID is only one of numerous technologies grouped under the term Automatic Identification (Auto ID), such as bar code, magnetic inks, optical character recognition, voice recognition, touch memory, smart cards, biometrics etc. Auto ID technologies are a new way of controlling information and material flow, especially suitable for large production networks. The RFID technology is means of gathering data about a certain item without the need of touching or seeing the data carrier, through the use of inductive coupling or electromagnetic waves. The data carrier is a microchip attached to an antenna (together called transponder or tag), the latter enabling the chip to transmit information to a reader (or transceiver) within a given range, which can forward the information to a host computer. The middleware (software for reading and writing tags) and the tag can be enhanced by data encryption for security-critical application at an extra cost. RFID is the technology of choice for object identification and tracking.

3. RFASSIST ARCHITECTURE AND FUNCTIONALITIES

We designed a low cost RFID based solution for blind people or visually impaired to recognize objects in a room. A block diagram of the system is shown in Fig. 2. This system consists of a RFID reader interfaced with microcontroller. The passive tags are placed on various objects such as laptop, books, DVD’s and other objects in the house or building. RFID reader fetches tag’s information and sends it to the microcontroller. This information can be used accordingly for different applications. For this purpose, three switches have been provided. When switch 1 is pressed, Microcontroller will play the corresponding audio file recorded for and unique to each object, residing in the audio circuit’s memory. The audio playback is relayed to microcontroller. Additionally, the corresponding object’s name is also displayed on the LCD. When switch 2 is pressed, tag’s information is serially transferred to the PC using serial RS232 communication interface. This would create a database in PC’s memory. It also records date and time of identification in the database. This database can be used later in library management system. When switch 3 is pressed, a gate connected to the microcontroller would open and a buzzer will also play when the correct tag is identified. After some time the gate will close automatically and again the buzzer will be heard. When some invalid card is detected, gates will remain close. This feature can be used in toll gates where RFID tags enabled car would not require any tickets. This will save lot of time and will ease traffic conditions.
This can also be implemented in security systems. When a valid card is identified only then gate is opened. The above system working has been represented by a flow chart, shown in Fig. 3.

Our RFID system comprises of an RFID reader working at 125 KHz and five passive RFID tags. These passive tags were placed on five different objects – Laptop, Mobile phone, CD, Remote Control and Book. The system utilizes an 8-bit PIC Microcontroller (PIC16F877A) especially designed for low-power operations. A further consideration is the selection of Microcontroller. Three different processors – AT89C51, PIC16F877A, and PIC18F4550 were considered and compared. The primary advantage of choosing PIC16F877A over AT89C51 mainly includes that it is a RISC design with low cost, inbuilt ADC, inbuilt comparator, powerful architecture, easy to use development tools like MPLAB IDE etc. PIC18F4550 is an advanced version of PIC16F877A. The former is suitable for our application.

In operation, the microcontroller receives the tag’s information from RFID reader on receiver pin. When switch 1 is pressed and a tag is detected the microcontroller identifies the tag using tag id. A prerecorded audio message is played corresponding to the tag using APR9600 interfaced with microcontroller. Also the object’s name is displayed on LCD. When switch 2 is pressed and a tag is identified, microcontroller transmits the tag’s information to the PC. A database is created on PC using software code written in C Programming language. The database also records time and date of identification. When switch 3 is pressed and a tag is identified a gate is opened using relay operating at 9V. The relay is driven using ULN2004A. A buzzer is also heard which provides aid to the blind. After some delay gate automatically closes and again a buzzer is heard.
Two types of communication are taking place in our system. One, in between the RFID reader and the microcontroller and other is in between the microcontroller and the PC. For both data transmission serial RS232 communication interface has been used. Both the transmissions occur at 4800 baud rate. The Hardware and Software architecture of the system is shown in Fig. 4 and 5, respectively. Fig. 6 shows the hardware of the system.

4. EXPERIMENTS AND RESULTS

The output of the system is object identification. By using RFID tags one can evaluate system’s performance. We are using 5 passive RFID tags. These RFID tags are placed on 5 different objects in an indoor environment- Remote Control, Book, CD, Laptop, Mobile. When a tag is detected, the three numbers printed on the tag is transmitted to the reader. One can use any of the three numbers to identify a tag.

When the number transmitted is same as that used by a microcontroller to identify an object, the following action takes place, if one of the three switches is pressed. Otherwise, no action takes place. When switch one is pressed the object’s name is displayed on LCD and a message is also audible which assist blind people in object identification.

When switch 2 is pressed the data from controller is transmitted to PC. PC displays the date and time of identification in real time and also stores the entire information as a database. The file formed here is a MS Excel file. The code for this is written in C language. The PC display and the database are shown in Figure 7 and 8 respectively.

When switch 3 is pressed, upon correct tag identification, gate opens and a buzzer is heard. After some time gate closes automatically and again a buzzer is heard. To demonstrate this, a CD loader is practically used (as shown in Fig. 6). The actions are also displayed on LCD for reference.
Fig. 5: Software Architecture of the system
Fig. 6: Full assembled PCB of RFASSIST (top view)

Fig. 7: Real Time display of date and time

Thus, the system is capable of identifying objects and the activities can also be stored in database for future referral. Undoubtedly it is very difficult for the blind to identify objects on their own. This system eases their work and provides audio messages which allow the identification of the object.

The prototype system presented here is a low frequency RFID system. This system was compared with an already existing low frequency RFID system. An RFID based attendance system is installed in BPIT College. Table I and Table II show the experimental results of the system as compared with existing RFID system.

<table>
<thead>
<tr>
<th>DATA</th>
<th>RFASSIST</th>
<th>ATTENDANCE SYSTEM</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>1.8 inch</td>
<td>1.2 inch</td>
</tr>
<tr>
<td>2</td>
<td>2 inch</td>
<td>1 inch</td>
</tr>
<tr>
<td>3</td>
<td>1.8 inch</td>
<td>0.8 inch</td>
</tr>
<tr>
<td>4</td>
<td>2 inch</td>
<td>1 inch</td>
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<tr>
<td>5</td>
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<td>1 inch</td>
</tr>
<tr>
<td>6</td>
<td>2 inch</td>
<td>1 inch</td>
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<tr>
<td>7</td>
<td>2.2 inch</td>
<td>0.8 inch</td>
</tr>
<tr>
<td>8</td>
<td>2.2 inch</td>
<td>1.2 inch</td>
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<tr>
<td>9</td>
<td>2 inch</td>
<td>1 inch</td>
</tr>
<tr>
<td>10</td>
<td>2 inch</td>
<td>1 inch</td>
</tr>
</tbody>
</table>

Table II: Comparison of Response Time of RFASSIST with existing Attendance system

<table>
<thead>
<tr>
<th>DATA</th>
<th>RFASSIST</th>
<th>ATTENDANCE SYSTEM</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>0.9 sec</td>
<td>0.9 sec</td>
</tr>
</tbody>
</table>
The average of the data shows that the attendance system was capable of identifying tags from a distance of less than 1 inch. On the other hand, RFASSIST can identify tags from a distance twice as that of the former i.e. approximately 2 inches. The response time was approximately same in both the cases.

Results show that the system is efficiently capable of object identification in short time for short distances. Also, the cost of developing the system was low as compared to others. The overall development cost of the system was Rs. 2500.

5. CONCLUSION AND FUTURE WORK

We presented a system that can easily identify objects for blind people using RFID, which would have been challenging for a vision only system. It makes use of low frequency RFID reader and passive tags. These tags operate wirelessly using power from RFID reader. They transfer the tag id to the reader and further to the microcontroller. Microcontroller makes use of this information to run various applications. It also displays object’s name on LCD. Moreover, Database was also created. This would help in library management. This system can also open and close gate which would be used in toll and security systems. We developed a prototype system for the above. This system is not suitable for a read range that is required in object identification and toll systems to identify the tags placed on vehicles. To implement such system in real world, we need to replace LF RFID reader with HF or UHF RFID readers.

This system shows up excellent performance when a tag is identified. However, when multiple tags are used simultaneously, the system hangs. Also, there should be a small delay between two adjacent tag identifications. This limits system performance. But, the application for which the system is designed doesn’t require simultaneous tag identification. The system is meant for blind people. They work at a slower pace as compared to us which provides sufficient delay between two events. Also, in toll systems when gate closes sufficient delay is there. When next vehicle enters the gate, the system is ready to sense another vehicle. Therefore, the above limitations do not affect our applications.

In future, the system would be capable of handling multiple tags simultaneously. Thus future design efforts will focus on improvements in simultaneous tag usage. The system would be used in future for location tracking as well. This would enable blind people to get location information and track objects. This would result in a complete package for the blind that could be used for multi applications.

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REFERENCES


A Novel Approach for Abnormality Detection of Macula and Fovea Region from Color Fundus Images

Shobhana.M and S.B.Chitrapreyanka

Abstract: The macula is the central portion of the retina, a small region rich in cones, the particular nerve endings that detect the color and upon which daytime vision depends. The macula is responsible for clear central vision. The remaining part of the retina is used for side or peripheral vision. The fovea centrals, usually known as the fovea, is located in the center of the macula region on the retina. The fovea is responsible for sharp central vision and is the most important part of the retina for human vision. Macular edema is often a complication of diabetic retinopathy and is the most common form of vision loss for people with diabetes particularly if it is left untreated. In this paper, a two stage methodology is employed to detect the macular edema and a simple, fast algorithm using mathematical morphology is employed to find the fovea region. An automatic disease detection system can significantly reduce the load of experts by limiting the referrals to those cases that require immediate consideration. The reduction in time and effort will be significant where a majority of patients screened for diseases turn out to be normal. Such a solution will be a value addition to the existing infrastructure of Diabetic Retinopathy (DR) screening.

Keywords: Abnormality detection, Diabetic Retinopathy, Fovea, Hard Exudates, Macular Edema.

I. INTRODUCTION

Diabetic Macular Edema (DME) is caused by leaking macular capillaries. There are many causes of macular edema. It is often associated with diabetes, where broken blood vessels in the retina begin to leak fluids, containing small amount of blood, into the retina. At times deposits of fats may leak into the retina. This leakage causes swelling in the macula region. Eye surgery, comprising cataract surgery, can increase the risk of developing macular edema due to blood vessels becoming irritated and leaking fluids. Macular edema that matures after cataract surgery is called Cystoid Macular Edema (CME).

Some of the other macular edema causes includes age-related macular degeneration, uveitis, retinal vein occlusion. Macular edema is often painless and may display few symptoms when it advances. When symptoms do follow, they are a sign that the blood vessels in the eye may be leaking. Symptoms of macular edema may include blurred or wavy central vision and/or colors appear washed out or changed. If left untreated, macular edema can cause severe vision loss and even blindness.

Macular edema occurs when fluid and protein deposits collect on or under the macula of the eye (a yellow central area of the retina) causing it to thicken and swell. The swelling may damage one's central vision, as the macula is present near the center of the retina at the back of the eyeball. Diabetic macular edema is mainly classified into focal and diffuse types. Focal macular edema which tends to leakage fluid is caused by foci of vascular abnormalities, primarily micro aneurysms, whereas diffuse macular edema is caused by dilated retinal capillaries in the retina [1], [2]. Two types of laser treatment for diabetic macular edema are focal and grid. The aim of focal laser treatment is to treat focal diabetic macular edema and to close leaking micro aneurysms. Similarly, the aim of grid laser treatment is to treat diffuse diabetic macular edema and is applied to areas of retinal thickening in which there is diffuse leakage to produce a retinal burn of mild to moderate intensity.

Diabetic Retinopathy is a complication of the retina due to diabetes and is a leading cause of blindness in urban population. Among the most serious DR abnormalities, hard exudates should be detected and treated immediately to prevent vision loss. Hard exudates occur in retina due to vascular deblities caused from prolonged diabetes. They are normally represented by bright yellowish lesions of different sizes and brightness. The location and extent of these lesions determine the resulting severity of diabetic retinopathy. Hard exudates are abnormal lesions caused by diabetic retinopathy in a diabetic’s eye. They are considered to be one of the bright intensity regions in the retinal images and appear as random yellowish patches. The size and distribution of exudates may vary during the progress of the disease.

DME is generally detected directly or indirectly. Direct ways are using stereoscopy or optical computed tomography images [3]. Indirect method is by detecting the presence of hard exudates (HE) in the retina. HE is formed due to secretion of plasma from capillaries resulting from the
complications of retinal vasculature and could lead to retinal swelling [5]. In color fundus images they appear as yellow–white deposits.

Fovea is the most important part of the retina for human vision. If the delicate cones of our fovea are destroyed, the person turns out to be blind. The size of fovea region in fundus image of the eye has relation with various diseases, which may lead to blindness. Usually the zone is approximated to a circle of radius 200 micron [1]. If the said radius is smaller then, we can conclude that there may be some deposition at the peripheral side, and that causes some infection or disease in eye, which may tend to retinopathy or blindness. Also the radius of the fovea region may indicate the stages of retinopathy [13].

Manual detection of fovea region by ophthalmologists is time consuming. Due to unavailability of trained ophthalmologists especially in developing countries like India, automation is highly needed. Fovea is characterized by the center of the macula. In fundus retinal image the macula is the darkest part approximated by a circle. Geometrically fovea is said to be located at a distance 2.5 times the diameter of the Optic Disk(OD) from its center [2]. In this paper, detecting the presence of hard exudates (HE) in macula region is considered a standard method to assess DME from color fundus images. In the next stage, the geometrical distance between OD and fovea region and the structure of the blood vessels is utilized to perfectly localize the fovea region.

II. ALGORITHM FOR DETECTING MACULAR EDEMA

![Figure 1: Block Diagram for Detection & assessment of Macular Edema](Image)

The macula is a dark structure roughly at the center of the retina. Hard Exudates which appear as clusters of bright lesions are usually well localized. In the absence of any HE (i.e., a normal retina), there is a rough rotational symmetry about the macula in the circular region of roughly twice the diameter of the optic disc.

We use this observation to derive relevant features to describe the normal and abnormal cases. Given a color fundus image, a circular region of interest (ROI) is first extracted and an intermediate representation also known as the motion pattern of the ROI is created [22]. Relevant features are then derived for to classify the given image as normal or abnormal.

A. Macula Localization (ROI Extraction)

Since the severity of DME is determined based on the location of HE clusters relative to the macula, the images acquired for DME detection usually focus around the macular region. We find the best fit circle within the fundus mask with macula at the center for a given image. The region within this circle is the desired ROI denoted as (see Fig.2 for an example).

![Figure 2: The region of interest centered on macula in a fundus image.](Image)

Detection of optic disc: An efficient detection of optic disc in color retinal images is a significant task in an automated retinal image analysis system. Its detection is prerequisite for the segmentation of other normal and pathological features. The position of optic disc can be used as a reference length for measuring distances in retinal images, especially for the location of macula [13]. In case of blood vessel tracking algorithms the location of optic disc becomes the starting point for vessel tracking.

The attributes of optic disc is similar to attributes of hard exudates in terms of color and brightness. Therefore it is located and removed during the hard exudates detection process, thereby avoiding false positives. In color fundus photograph, optic disc appears as a bright spot of circular or elliptical shape, interrupted by the outgoing vessels [8]. It can be seen that optic nerves and vessels emerge into the retina through optic disc. It is situated on the nasal side of the macula and it does not contain any photoreceptor. Therefore it is also called the blind spot.

Localization of Optic Disc: The localization of optic disc is important for two purposes. First, it serves as the baseline for finding the exact boundary of the disc. Secondly, optic disc center and diameter are used to locate the macula in the image [10]. In a color retinal image the optic disc belongs to the brighter parts along with some lesions. The central portion of disc is the brightest region called optic cup, where the blood vessels and nerve fibers are not present. A threshold is to be applied, that will separate part of the optic disc and some other unconnected bright regions from the background.

Elimination of Vessels: The optic disc region is usually fragmented into multiple sub-regions by blood vessels that have comparable gradient values. A homogeneous optic disc region is needed for segmentation using geometric active
contour algorithm. Median filter with appropriate size is used to remove interfering blood vessels from the optic disc region resulted in heavy blurring of disc boundaries [5]. Instead a better result is achieved with gray level mathematical morphology to remove irrelevant vessels from the optic disc region.

Gray scale mathematical morphology provides a tool for extracting geometric information from gray scale images. A structuring element is used to build an image operator whose output depends on whether or not this element fits inside a given image. Shape and size of the structuring element is chosen in accordance with the segmentation task. The two fundamental morphological operations are dilation and erosion [5]. Due to dilation operation the small interfering blood vessels are detached. Next, erosion is done to restore the boundaries to their former position.

Detection of Macula: The macula is a depression in the center of macular region and appears as a darker area in a color retinal image. It is located temporal to the optic disc and has no blood vessels present in its center.

**Figure 3:** Illustration of finding a search area to localize macula.

Since the location of macula varies from individual to individual, a rectangular search area has to be defined. In a standard retinal image the macula is situated about 2 disc diameter (DD) temporal to the optic disc [17], [18]. Based on this prior knowledge a rectangular search area is formed as shown in Fig.3. The width of the search area is taken equal to two Disc Diameter (2DD) as the mean angle between the macula and the center of the optic disc to the horizontal. A small pixel window of size 40x40 is formed to scan the entire area and the average intensity at each pixel location is calculated. The center of the window having the lowest average intensity is taken as the center of the macula. Fig.3 shows the result of automatic macula detection method. As the macula is localized, the whole macular region can be determined for detecting the presence or absence of hard exudates.

**B. Combining the Motion Patterns**

When an object in a scene moves at a high speed, it usually leaves a smearing pattern in the captured image. Generally, the spatiotemporal changes recorded by the sensor are characteristic of the moving object. Signal aggregation at sensor locations in human eyes and camera, gives rise to the smearing effect. For simulating this effect, a motion is induced in a given image to generate a sequence of images. These are combined by applying a function to coalesce the intensities at each sensor (or pixel) location (i.e. pixel position across the sequence) to give rise to a motion pattern [19]. Let a region of interest (ROI) in a medical image be denoted as $I(r)$. By inducing motion on image $I$, a motion pattern $I_{MP}$ is obtained as follows

$$I_{MP}(r) = f(G_N(I(r)))$$

Where $r$ denotes a pixel location, is a transformation representing the induced motion which is assumed to be rigid. $G_N$ generates $N$ transformed images which are combined using $f$ to coalesce the sampled intensities at each pixel location as given in (1). Since, the severity of the disease is directly related to the radial distance of HE in the circular ROI, rotational motion is induced to generate the desired. The transformation function is applied to generate a sequence of images which are rotated versions. The spatial extent of smearing of intensities depends on the maximum rotation whereas the sampling rate at each location is directly related to the size of each rotation step. Consider a disk with a single circle near the periphery modeling a lesion. When rotation is applied to this pattern, a set of patterns are generated. When two patterns will be generated and their union is the second pattern [18], [19]. The remaining patterns are the result of the union of patterns generated with decreasing step size. It can be observed that a decrease in the step size results in several copies of the lesion in the final result. Here, the motion pattern is obtained by using the union operation as the coalescing function.

Accordingly, two functions namely mean and maximum were considered in this work. While the coalescing function mean obtained tries to achieve the averaging effect observed in motion blur, maximum obtained tries to exploit the fact that HE usually appear brighter than any other structures in the background at the same radial distance [6]. The original and motion pattern images illustrate the effect of the two coalescing functions on a normal and two abnormal fundus images.

**C. Feature Selection**

The motion pattern generated by inducing motion on image $I$, results in the smearing of lesions when present, along the motion path. To effectively describe this motion pattern, we propose to use a descriptor derived from the Radon space [10]. The Radon transform of is the integral of along a line oriented at and distance from the origin.

$$P \propto(r) = f(x,y)\delta(r-x\cos \propto - y\sin \propto)dx\,dy$$

Where, $\propto$ is the angle between the line and y axis.

The image is projected to obtain a vector response for every angle and the desired feature vector then is derived by concatenating the responses for different orientations. The spatial extent of any HE that may be present is enhanced in
the motion pattern and is in turn reflected immediately in the projection based feature vector [12]. Thus, the feature vector for an abnormal retina will have several peaks in its profile due to intensities corresponding to HE. On the other hand, the feature vectors for a normal retina will have relatively uniform values resulting in a compact normal subspace. These feature vectors are used for learning the subspace corresponding to normal images.

D. Abnormality Detection

Single class classifiers are used for learning normal cases. In this approach, a classification boundary is formed in the feature space around the subspace corresponding to normal cases [15]. If a new image is transformed to this feature space, it is within this boundary, then it is classified as normal and abnormal elsewhere. Two class classifiers are used in this work are: Gaussian data description (Gaussian DD) and principal component analysis data description (PCA DD).

Gaussian Data Description: Here, the normal class is modeled as a Gaussian distribution. The model parameters, namely, the mean $\mu$ and the covariance $\Sigma$ are computed for the training set made of normal cases. Classification of a new case is based on the Mahalanobis distance between the new case and the normal subspace.

$$D(g(I_{MP})) = (g(I_{MP} - \mu)^T \Sigma^{-1}(g(I_{MP} - \mu))$$  \hspace{1cm} (3)

Principal Component Analysis: Here, a linear subspace is defined for the normal cases. The eigenvectors corresponding to the covariance matrix of the training set is used to describe the subspace. The feature vector for a new case is projected to this subspace and again reconstructed.

For both the above single class classifiers, the classification between normal and abnormal images is then performed using an empirically determined threshold on and for Gaussian DD and PCA DD classifiers, respectively[8]. Thus far, we have described the methodology for determining if a given image is normal or abnormal.

E. Severity Assessment of Macular Edema

Assessing the severity of macular edema is very important. The macula in a normal image is relatively darker than other regions in the fundus image and is characterized by (rough) rotational symmetry. We use this symmetry information to establish the risk of exhibiting edema [13], [15]. Good degree of symmetry is taken to indicate the abnormality is not inside macula and hence it is declared as a moderate case. Asymmetry of the macula on the other hand implies abnormality is within the macula and hence the case is considered as severe.

A threshold on the symmetry measure $S$ is used for assessing the degree of abnormality of an image as moderate or severe risk of DME [9]. Let $S_{max}$ and $S_{min}$ be the maximum and minimum symmetry values for normal images in the training set used for abnormality detection. Then the severity of a given abnormal image is determined by comparing the symmetry measure of this image $S(I_a)$ against a threshold $T$ as follows,

$$\text{Severity}(I_a) = \begin{cases} \text{moderate; if } S(I_a) \leq T \\ \text{severe; otherwise} \end{cases}$$  \hspace{1cm} (4)

Threshold $T$ should be selected to be a percentage $p$ of the maximum symmetry value for normal images. Hence, the threshold is selected as

$$T = P(S_{max} - S_{min}) + S_{min}$$  \hspace{1cm} (5)

III. ALGORITHM FOR FOVEA EXTRACTION

Input: Gray-scale fundus image ($I_1$), an image $I_S$ (contains only blood vessels), approximate center (G) and diameter (d) of the optic disk.

Output: Macula and fovea region.

Step 1: Locate a point P horizontally at a distance $2.5 \times d$ from G towards the centroid.

Step 2: A vertical strip of width k pixels is considered around P perpendicular to GP.

Step 3: Apply a k\times k sliding window along the strip and form the chain of numbers denoting the black pixels in the window.

Step 4: Find the maximum run length of zeros, L in the number chain.

Step 5: Let S and E are the start and end position corresponding to L and D be the mid position of S and E.

Step 6: Consider a binary image $BW$ of size same as the input image with only a black pixel at position D. Dilate $BW$ by a disc of radius DS to obtain $BW_d$.

Step 7: Obtain portion of the gray-scale image as R,$I_1$ corresponding to the black region in $BW_d$.

Step 8: Binarize the gray scale image to approximate macula region [16].

Step 9: Refine binarized image by removing noise and fitting the circle to obtain final macula region.

Step 10: Fovea region is to be detected as small area around the center of macula. It is marked by a red colored circle.

IV. SIMULATION OUTPUT

The simulation tool used for processing input color fundus images is MATLAB. The version of MATLAB is 7.12. Image processing toolbox is used for simulation. The input contains both normal and abnormal images. The input eye images are in digital format. It is in .jpg format.

A. Conversion to Gray Scale Image

To reduce the correlated color information in color fundus image as shown in Fig 4(a), RGB image is converted into gray-scale as input image as shown in Fig 4(b).
B. Morphological Operations on Gray Scale Image

Morphological opening operation and Morphological closing operation are applied to a disk shaped structuring element on gray-scale image to reduce the small noise and to remove the vessels structure as shown in Fig.5(a) and Fig.5(b) respectively.

Due to dilation operation, the small interfering blood vessels are detached. This will make the input image slightly blurred. Next, erosion is done to restore the boundaries to their former position.

C. Blood Vessel Extraction and Optic Disc Detection

The blood vessels of the disk are found using MATLAB morphological filters. Based on the adaptive mathematical morphology, the origin of the optic disc is identified.

D. Generation of Motion Patterns

When rotation is applied to the disk pattern in steps of \( \theta_o = \pi \), a set of patterns is generated. When two patterns will be generated and their union is the second pattern. The remaining patterns are the result of the union of patterns generated with decreasing step size.

The motion pattern is obtained by using the union operation as the coalescing function. As shown in Fig.7 similar eight motion patterns are generated in the decreasing rotation steps.

E. Combined Motion Patterns and Segmented Image

Combining the eight motion by applying the rotation steps in decreasing order from (0 to 2 \( \pi \)) is obtained as shown in the Fig.8(a) and the segmented image with affected areas are obtained as shown in the Fig.8(b).

F. Fovea extraction and the image with affected areas

A point is located horizontally at a distance 2.5x from optic disk center towards the centroid. A kxk sliding window is applied along the strip and forms chain of numbers denoting the black pixels in the window.

Determining the severity of macular edema is another task. The macula in a normal image is relatively darker than other regions in the fundus image and is characterized by rotational symmetry. Good degree of symmetry is taken to indicate the abnormality is not present inside macula and hence it is declared as a moderate case. Asymmetry of the macula on the other hand implies abnormality is within the macula and hence the case is severe.

Figure 4: (a) color fundus image  (b) gray scale image

Figure 5: (a) erosion operation  (b) dilation operation

Figure 6: (a) Blood vessel extraction   (b) OD detection

The macula, which is a depression in the center of macular region, appears as a darker area in a color retinal image. It is located temporal to the optic disc and has no blood vessels present in it. Blood vessel extraction for the input image (vessel detected image) is shown in Fig.6 (a) and the optic disc detected is shown in Fig.6 (b).

Figure 7: Motion patterns of decreasing rotation steps

Figure 8: (a) combined motion pattern (b) segmented image

Figure 9: (a) fovea localization  b) affected areas

Maximum run length of zeros is found in the number chain. That is known as a fovea region. It is marked by a red colored circle as shown in fig.9(a) and checking of that image to spot the disease is done by visualizing the change in the
shape of fovea. The affected areas of the retinal image are shown in fig.9(b).

V. RESULTS AND DISCUSSION

The output segmented image for determining the macula edema, that is detecting the presence of hard exudates near the macula region is obtained through different steps in the proposed algorithm. The steps includes region of interest extraction, generation of motion patterns, feature selection, abnormality detection, severity analysis. The following are observed in the command window of MATLAB: the elapsed time, normal or abnormal retina and if it is abnormal, whether it is moderate or severe risk. Then the fovea region is extracted and depending on the change in shape of the fovea, the affected regions are found out.

VI. CONCLUSION

DME detection and assessment provides significant contributions which include a hierarchical approach to the problem, a novel representation for the first level, to classify an image as normal or abnormal (containing HE) and a rotational asymmetry measure for the second level, to assess the severity of DME. This novel representation captures the global image characteristics. Such global features have not been used successfully earlier for HE detection. In the first level, a supervised technique based on learning the image characteristics of only normal patients is used for detecting the abnormal cases pertaining to HE. In the second level, the severity of the abnormality is assessed by analyzing the rotational asymmetry of the macular region in retina. Further in this paper, a new efficient method is described to localize the fovea in retinal fundus image. Morphological operators and geometrical features are used to localize the fovea region successfully. Proposed scheme is simple but efficient in extracting the fovea region. The extracted macula and fovea region may help in further diagnosis of eye related diseases.

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A QosBased Performance Evaluation of Wireless Networks using OPNET modeler

Vandana T. Bhatt and Abhishek Bhatt

Abstract- WLANs are being used for military, multimedia and health application, where high system performance and the ability to stay in link is extremely required. It can be considered the wireless version of Ethernet, which supports best-effort service. We have measured the performance of next generation wireless networks in terms of delay, media access delay, throughput, retransmission attempts. These are essential metrics which are used to identify the performance of a given wireless networks. Based on these measurement, the decision for networks selection is made i.e. Using vertical handover or alternate handover, user’s call/connection is seamlessly transferred to good QoS enabled network.

In this paper the performance optimization methods have been presented using an advanced networks simulator, OPNET modeler 14.0. Further effect of segmentation on delay, throughput and mobility is analyzed. Finally a utility table is provided from calculation drawn from six different scenarios.

In adhoc networks, the throughput is increased when segmentation is enabled with a subsequent increase in delay. Thus this network will perform better for networks with high tele-density. ESS performs better than infrastructure BSS networks as throughput is increased and delay is reduced. This network can be recommended for video conferencing and high speed data transmission. For internet application in infrastructure BSS, PCF/DCF performs better than DCF. Throughput is increased and delay is increased. Finally the results are compiled in a utility measurement table.

Keywords-OPNET MODELER 14.0,QoS,Throughput, WLAN

I. INTRODUCTION

The best-known WLAN standard is IEEE 802.11, which has several supplementary standards. The legacy IEEE 802.11 was introduced with carrier sense multiple access/collision avoidance (CSMA/CA) MAC protocol and three different physical layer mechanism: direct sequence spread spectrum (DSSS), frequency hopping spread spectrum (FHSS), and infrared (IR). Since then, the standard has been enhanced with two physical layer standards: IEEE 802.11b and IEEE 802.11a. IEEE 802.11b uses High rate DSSS (HR/DSSS) and IEEE 802.11a uses orthogonal frequency division multiplexing (OFDM) [12]. The IEEE 802.11e MAC protocol is expected to be ratified for providing quality of access. With state-of-the-art simulators (OPNET), throughput performance can be measured in MAC layer.

II. RELATED WORK

In [1], the overall performance of the IEEE 802.11 Wireless Local area networks has been analyzed in detail with the help of OPNET Modeler. The parameters like throughput, media access delay, the number of retransmission attempts, dropped data packets etc. for data rate, fragmentation threshold, RTS/CTS threshold, physical characteristics and the buffer size.

In [2], the performance of WLAN has been analyzed based on the parameters of response time, bandwidth,physical characteristics, roaming capability and the access methods.

In [3], the authors investigated how to improve voice and data service support in the cellular/WLAN integrated network by applying admission control. It is observed from the numerical results that the resource utilization can be maximized when a balance is achieved in distributing the voice and data traffic load to the overlaying cell and WLAN.

In [4], Kotz and Essein characterize campus-wide wireless network usage at Dartmouth college, focusing on infrastructure mode using access points.

In [5], Balachandran report on network performance and user behaviour for general internet access by San Diego. They find that for this set of technology-literate users a wide range of internet application are used, user behaviours are diverse, and overall bandwidth demands are moderate.

In [6], the authors have investigated some of the factors affecting throughput and delay performance of IEEE 802.11 WLANs in the PCF mode as compared to the DCF mode. and they agree that PCF behave better in terms of throughput when all nodes have data to send.

In [7] research is done to study, the Point Coordination Function (PCF) of IEEE 802.11, the Enhanced Distributed Coordination Function (EDCF) of the proposed IEEE 802.11e extension to IEEE 802.11. The metrics used were Throughput, Data Drop, Retransmission and Medium Access Delay.

The results showed that the performance of EDCF was better in providing QoS for real-time interactive services (like video conferencing) as compared to DCF Whereas the DCF’s overall performance was marginally better for all.
kinds of services taken together. In our own work the performance of WLAN has been analyzed. Throughput, Media Access Delay, Delay, Retransmission Attempts and Data Dropped are taken as performance metrics

III. SIMULATION SCENARIOS

In our work, we use OPNET Modeler 14.0 [8] to model a WLAN. We have taken five different scenarios to study the performance of WLAN. We have taken 300 simulation seconds to carry out the simulation.

Scenario 1 – Adhoc N/w with varying no. of users
We model an infrastructure WLAN. The network consists of a four fixed nodes without any access point. So all the workstation can directly communicate with each other.

### TABLE 1

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Throughput (Mbps)</th>
<th>Delay (ms)</th>
<th>Media Access Delay (ms)</th>
<th>Data Dropped (Mbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Segmentation Disabled</td>
<td>1429485</td>
<td>0.00144</td>
<td>0.00065</td>
<td>743519</td>
</tr>
<tr>
<td>Segmentation Enabled</td>
<td>2113434</td>
<td>0.0044</td>
<td>0.0034</td>
<td>Zero</td>
</tr>
</tbody>
</table>

Analysis of scenario 1 simulation results

1. By increasing the no. of users beyond 8 reduces the throughput.
2. Media Access delay and Delay increases with the no. of nodes.
3. Throughput and load are exactly same for a particular scenario.
4. For a 100mX100m office, WLAN can afford max 8 no. of users effectively.

**Scenario 2 – Independent BSS with segmentation Enabled**

We model an infrastructure WLAN. The network consists of a four fixed nodes without any access point. Hero one attribute is “Packet Generation Argument” whose sub-attribute is “Segmentation size (bytes)”. If set to “No Segmentation”, then each generated packet is immediately sent to the lower layer whose size is determined based on the value of the “Packet Size” attribute. In this scenario, we have set the segmentation size to 1500.

Simulation results for scenario 2

![Figure 1. Delay Vs Time](image1)

![Figure 2. Media Access Delay](image2)

![Figure 3. Throughput Vs Time](image3)

![Figure 4. Data Dropped Vs Time](image4)

![Figure 5. Throughput Vs Time](image5)
TABLE 2
Analysis of scenario 2 simulation results

<table>
<thead>
<tr>
<th>No. of Users</th>
<th>Throughput (Mbps)</th>
<th>Delay (Sec)</th>
<th>Media Access Delay (Sec)</th>
</tr>
</thead>
<tbody>
<tr>
<td>4</td>
<td>1429485.512</td>
<td>0.00135</td>
<td>0.000580</td>
</tr>
<tr>
<td>8</td>
<td>2967878</td>
<td>0.02216</td>
<td>0.0206</td>
</tr>
<tr>
<td>10</td>
<td>2876350</td>
<td>0.37457</td>
<td>0.3688</td>
</tr>
<tr>
<td>12</td>
<td>2799374</td>
<td>0.47816</td>
<td>0.4716</td>
</tr>
</tbody>
</table>

1. Throughput is increased when segmentation is enabled.
2. Delay and Media Access Delay also increases up to 4 times.
3. Data Dropped is reduced to zero.

Scenario 3 –
In this scenario we have done the comparison b/w Adhoc and Infrastructure BSS
So far we have worked on ad hoc N/W. Now we will take an infrastructure BSS. Then comparison between these two modes is done.

Simulation results for scenario 3

TABLE 3
Analysis of scenario 3 simulation results

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Throughput (Mbps)</th>
<th>Delay (ms)</th>
<th>Media Access Delay (s)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ad hoc mode</td>
<td>187986</td>
<td>0.027</td>
<td>0.0307</td>
</tr>
<tr>
<td>Infrastructure mode</td>
<td>193325</td>
<td>1.39</td>
<td>9</td>
</tr>
</tbody>
</table>

1. Throughput slightly increases with infrastructure BSS as compared to Adhoc N/W.
2. But Delay and Media Access Delay both increases largely as compared to ad hoc network.

Scenario-4
Infrastructure BSS With DCF/PCF Mode
This scenario has eight wireless LAN-based workstation in a simple network configuration (infrastructure BSS) which demonstrates the access method used by the wireless LAN [3].
PCF provides a contention-free (CF) frame transfer. The medium access during the CF is regulated by the point coordinator (PC) which resides in the access point (AP).
Statistics Studied:
The number of retransmission for a PCF enabled station is very less compared to a non PCF node. Also, the throughput will be much higher for a PCF enabled compared to a non PCF node with a similar load. This is because PCF enabled
node will be able to transmit during both CFP and CP.

**Simulation results for scenario 4**

![Figure 9. Media Access Delay Vs Time](image)

![Figure 10. Retransmission Attempts Vs Time](image)

![Figure 11. Throughput Vs Time](image)

**TABLE 4**

Analysis of scenario 4 simulation results

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Throughput (kbps)</th>
<th>Media Access Delay (ms)</th>
<th>Retransmission Attempt (Packets)</th>
</tr>
</thead>
<tbody>
<tr>
<td>DCF</td>
<td>287986</td>
<td>0.0021</td>
<td>4.656</td>
</tr>
<tr>
<td>DCF_PCF</td>
<td>294407</td>
<td>0.00424</td>
<td>1.165</td>
</tr>
</tbody>
</table>

Analysis of scenario 4 simulation results
1. Throughput is more in PCF than DCF mode due to contention mode and polling method alternatively.
2. But Delay and Media Access Delay is less in DCF mode by method of avoiding collision.

Scenario-5

Wireless LAN Developed With Internet In DCF/PCF Mode

The scenario consist of a wireless and a wireline network. The purpose of the scenario is to demonstrate the inter-communication between the wireless and wireline network through the internet back-bone. The site 1 and site 2 subnet each contain 07 wireless station; all station comply with the wireless LAN (802.11) protocol. The access point nodes in site 1 and site 2 connect each subnet to the wireline network. The clients in the wireless LAN are trying to communicate with servers at the remote site via IP clouds. Parameters configuration of WLAN station, others parameters are default. This paper take two types of access mechanism of DCF and DCF/PCF as scenery to simulate in order to compare and analysis results, the simulation time is 10 minutes.

**Simulation results for scenario 5**

![Figure 12. Data Dropped Vs Time](image)

![Figure 13. Throughput Vs Time](image)

![Figure 14. Media Access Delay Vs Time](image)
TABLE 5
Analysis of scenario 5 simulation results

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Throughput (kbps)</th>
<th>Delay (ms)</th>
<th>Media Access Delay (ms)</th>
<th>Data Dropped (kbps)</th>
</tr>
</thead>
<tbody>
<tr>
<td>DCF mode</td>
<td>3678366</td>
<td>0.92</td>
<td>0.9138</td>
<td>7670728</td>
</tr>
<tr>
<td>PCF/DCF mode</td>
<td>4754777</td>
<td>7.12</td>
<td>4.926</td>
<td>7325401</td>
</tr>
</tbody>
</table>

1. DCF access mechanism reduces the delay by method of avoiding collision because it uses the mechanism CSMA/CA
2. Throughput of WLAN is higher by using DCF/PCF access mode than DCF access mode because DCF/PCF access mode adopt contention mode and polling mode alternately whereas DCF mode uses only the contention mode.

IV. CONCLUSION

In this, several methods for improving WLAN performance were investigated. Using OPNET software tool for network management and capacity planning several network models were created, different scenarios were chosen, simulation were executed and results were viewed and analyzed. Our analysis indicates that tuning the MAC layer characteristics related parameters, such as DCF/PCF, Fragmentation threshold improves the network performance.

We have simulated throughput, media access delay, retransmission attempts, load, delay etc. as quality of service measures for WLAN and adhoc network.

We have classified over simulation in 5 different scenario and their conclusion are as follows:-

[1] In adhoc networks, delay and throughput obtained for 8 users is optimum.
[2] In adhoc networks the throughput is increase when segmentation is enabled with a subsequent increase in delay. Thus this network will perform better for networks with high tele-density.
[3] DCF performs better in infrastructure BSS networks as compared PCF/DCF mode.
[4] BSS performs better than adhoc network as throughput id increase and delay is reduced by 10 times. Thus this network can be recommended for video conferencing and high speed data transmission.
[5] For internet application in infrastructure BSS, PCF/DCF performs better than DCF. Throughput is increased an d delay is decreased thus this is recommended for large scale networks.

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Performance Evaluation of Artificial Neural Network Classifier Based On Receiver Operating Characteristic (ROC) Curves

Ms. J.D. Dhande and Dr. S. M. Gulhane

Abstract: Classification is one of the most active research and application area of neural networks. Artificial Neural Networks have been applied as a classifier to find one “best” detection rate using Receiver operating characteristic (ROC) curve. This research work has been proposed the design of multilayered neural network (MLP NN) classifier for classification of Echocardiogram dataset as a two class problem. The performance of MLP NN classifiers are examined by generating the ROC curves for training and testing dataset. ROC gives better results in terms of Area under the ROC curve (AUC) is greater and in the sense of being composed of a better distribution of operating points. Simulation results shown that the classifier can achieve more than 90% classification accuracy using ROC curves.

Keywords: Artificial Neural Network (ANN), Classifier, MLPNN, Receiver operating characteristic (ROC)

I. INTRODUCTION

Classification is one of the most frequently used decision making tasks of human activity. A classification problem occurs when an object needs to be assigned into a predefined class or group based on a number of observed attributes related to that object. Many problems in business, science, industry, and medicine can be treated as classification problems. Examples, includes bankruptcy, prediction, credit scoring, medical diagnosis, Quality control, hand written character recognition and speech recognition.

Traditional statistical classification procedure such as discriminate analysis is built on the Bayesian decision theory [1][5]. The effectiveness of these methods depends on a large extent on the various assumptions. Artificial neural networks have emerged as an important tool for classification. The recent vast research activities in neural network classification have established that artificial neural networks are a promising alternative to various traditional classification methods. The neural networks are nonlinear models, which makes them flexible in modeling real world complex relationships. In spite of the successful application of ANN’s to several pattern recognition problems [1][2], ANN also provide a powerful tool to help doctors to analyze, model and make sense of complex clinical data across a broad range of medical applications [3]. Most applications of artificial neural networks to medicine are classification problems; that is, the task is on the basis of the measured features to assign the patient to one of a small set of classes [4][5]. The performance of classifier is measured in terms of classification accuracy and another method of specifying the performance of classifier for generating the Receiver Operating Characteristic (ROC) Curve.

II. BACKGROUND OF RECEIVER OPERATING CHARACTERISTIC (ROC) CURVE

The Receiver Operating Characteristic (ROC) curve is the method of specifying the performance of a classifier is to note its true positive (TP) rate and false positive (FP) rate for a dataset. The True positive (TP) rate is the percentage of target samples that are correctly classified as target samples. The false positive (FP) rate is the percentage of non target samples that are incorrectly classified as target samples. For particularly applications, we may require the...
classifier to operate at some point other than the one to which it naturally trained.

An ROC curve is a plot of operating points showing the possible trade-off between a classifier’s TP rates versus its FP rate. The TP rate is commonly referred to as “sensitivity” and (1-FP) rate is called “specificity”. The ROC curves are a useful way to interpret sensitivity and specificity levels and to determine related cut scores. ROC curves are a generalization of the set of potential combinations of sensitivity and specificity possible for predictors. ROC curve analyzes not only provide information about cut scores, but also provide a natural common scale for comparing different predictors that are measured in different units. An overall indication of the diagnostic accuracy of a ROC curve is the area under the curve (AUC). Area under the curve values are closer to 1 indicate the screening measure reliably distinguishes among satisfactory and unsatisfactory performance, whereas values at .5 indicate the predictor is no better than chance. Figure 1 shows the two typical ROC curves for performance evaluation.

![Figure 1 Two typical ROC Curves](image)

Receiver Operating Characteristic (ROC) curve analysis is a method of measuring diagnostic performance in medical application. This work examines the method of performance evaluation of ANN classifier by generating ROC curve for two class problem.

### III. ARTIFICIAL NEURAL NETWORK CLASSIFIER

An Artificial Neural Network (ANN) is a computational model that attempts to account for the parallel nature of human brain. An ANN is a network of highly interconnecting processing elements (neurons) operating in parallel. These elements are inspired by biological nervous systems. As in nature, the connections between elements largely determine the network function. A subgroup of processing elements is called a layer in the network. The first layer is the input layer and the last layer is the output layer. Between the input and output layer, there may be additional layers of units called hidden layers. Figure 2 represents the typical neural network. The neural network is trained to perform a particular function by adjusting the values of the connection weights between elements.

![Figure 2 A typical Neural network](image)

The most popular neural network is Multi-layer feed forward network (MLP NN) with back-propagation learning algorithm. Although many types of neural networks can be used for classification purposes [8-9]. The main focus on multi-layer perceptron neural network which are the most widely studied and used neural network classifier. The back propagation neural network architecture is the most popular, effective and easy-to-learn model for complex multilayer networks. To this network, training inputs are applied to the input layer of the network, and desired outputs are compared to the output layer. During learning process, a feed forward sweep is made through the network, grid the output of each element is computed layer by layer. The difference between the output of the final layer and the desired output is back-propagated to the previous layers, usually modified by the derivative of the transfer function, and the connection weights are normally adjusted. This process proceeds for the previous layers until the input layer is reached [5][9].

### IV. METHODOLOGY AND SIMULATION RESULTS

The echocardiogram dataset is used for design the ANN based classifier. This section gives Overview of echocardiogram dataset, training of neural network, and
simulation result of neural network design with performance curves.

A) Dataset Description
In this experiment the medical data related to echocardiogram dataset of patients for survival analysis after the heart attacks is considered. This dataset has been taken from publically available UCI repository machine learning [7]. The dataset concerns classification of patient’s survival at least one year that is death or life after a heart attack. The dataset size is reduced by ignoring one attribute values from given dataset. Table 1 gives attributes description of echocardiogram dataset. The dataset is partitioned into training set having 100 samples and testing set having 32 samples for design and test the classifier.

Data Representation:

Number of instances: 132
Number of attributes: 11 and a class attribute
Class:
Class 0: A patient was either dead after 1 year or had been followed for less than 1 year.
Class 1: Patients was alive at 1 year

Table 1 Attributes Description

<table>
<thead>
<tr>
<th>Sr. no</th>
<th>Attributes</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Survival</td>
<td>Number of months patients survived</td>
</tr>
<tr>
<td>2</td>
<td>Still-alive</td>
<td>A binary variable: 0 = dead at end of survival period, 1 means still alive</td>
</tr>
<tr>
<td>3</td>
<td>Age <em>at</em> heart attack</td>
<td>Age in years when heart attack occurred</td>
</tr>
<tr>
<td>4</td>
<td>Pericardial effusion</td>
<td>Binary: 0 means no fluid, 1=fluid</td>
</tr>
<tr>
<td>5</td>
<td>Fractional shortening</td>
<td>A measure of contractility around the heart lower numbers are increasingly abnormal</td>
</tr>
<tr>
<td>6</td>
<td>Epss</td>
<td>E-point septal separation, another measure of contractility</td>
</tr>
<tr>
<td>7</td>
<td>Lvm</td>
<td>A measure of the size of the heart at end–diastole</td>
</tr>
<tr>
<td>8</td>
<td>Wall-motion-score</td>
<td>A measure of how the segments of the left ventricle are moving</td>
</tr>
<tr>
<td>9</td>
<td>Wall-motion-index</td>
<td>Equals wall motion</td>
</tr>
</tbody>
</table>

10 Mult A derivate var
11 Name Name of patients

A) Training the Neural Network
In this experimentation, Multilayer feed forward neural network is created by using the training set and the network is trained on given data and used back propagation learning algorithm. Multilayer feed forward neural network is three layers model. The input layer network consists of 11 neurons to represent each attributes as the database attributes. The numbers of classes are two: class 0 and class 1. The output layer consists of one neuron to represent these classes. The hidden layer consists of 20 neurons in MLP neural network. The design model of MLPNN classifier is 11-20-1. The optimal value of hidden node is obtained based on trail and error experimentation. This model is trained using back propagation learning algorithm with more runs. Figure 3 shows the relation between the number of epochs and the performance in term of Mean square error (MSE) during training process for MLP NN network model. For training the MLPNN 11 epochs required with minimum MSE value.

![Figure 3 Training curve for MLPNN](image)

B) Performance of Neural Network
MLPNN model is trained on 11 epochs and this network is validated on testing set. The testing set with 32 samples is tested on trained network. For performance evaluation of network, Receiver operating characteristic (ROC) curves is plotted for training and testing dataset. ROC curve for training data with better classification curve with best TP and FP rates has shown in figure 4.
For generalization of trained network, the performance evaluated on testing set as a unknown input samples as input to trained network and output of network is calculated with the adjusted weights. The output of network is compared with the target output to study the learning ability of network for classification of echocardiogram dataset with better accuracy based on trail and error experimentation. ROC curve is plotted for testing dataset and is shown in figure 5.

Figure 5 ROC curve for MLPNN testing set

V. CONCLUSION

For wide range of applicability of ANN and their ability to learn complex and non linear relationships including noisy or less precise information, neural networks are well suited to solve problems in biomedical field. In this study, neural network technique is adopted for classification of medical dataset with back propagation learning algorithm. The performance of network is analyzed on training and testing dataset with ROC curve generating from trained classifier. The manipulation of classifier parameters will in turn move a decision boundary in feature space between the two classes and result in a new sensitivity/specificity tradeoff. We examined that the performance of design neural network classifier by generating the better ROC curve in sense of having greater area under the ROC curve (AUC) and achieved more than 90% classification accuracy on generalization.

REFERENCES

GSM Based Gas Leakage Detection System

Ashish Shrivastava, Ratnesh Prabhaker, Rajeev Kumar and Rahul Verma

Abstract: Gas leakage is a major problem with industrial sector, residential premises and gas powered vehicles like CNG (compressed natural gas) buses, cars. One of the preventive methods to stop accident associated with the gas leakage is to install gas leakage detection kit at vulnerable places. The aim of this paper is to present such a design that can automatically detect and stop gas leakage in vulnerable premises. In particular gas sensor has been used which has high sensitivity for propane ($\text{C}_3\text{H}_8$) and butane ($\text{C}_4\text{H}_{10}$). Gas leakage system consists of GSM (Global System for mobile communications) module, which warns by sending SMS. However, the former gas leakage system cannot react in time. This paper provides the design approach on both software and hardware.

Keywords: GSM (Global System for mobile communications), CNG (compressed natural gas), LPG (Liquefied petroleum gas), Gas sensor MQ-6, stepper motor Driver IC (ULN2003A), Microcontroller (AT89C51), LCD (Liquid crystal display), RF (Radio Frequency) link, Decoder HT12D, Encoder HT12E.

I. INTRODUCTION

LPG consists of mixture of propane and butane which is highly flammable chemical. It is odorless gas due to which Ethanethiol is added as powerful odorant, so that leakage can be easily detected. There are other international standards like EN589, amyl mercaptane and tetrahydrothiophene which are most commonly used as odorants. LPG is one of the alternate fuels used now days. Sometimes liquefied petroleum gas is also known as LPG, LP gas, Auto gas etc. This gas is commonly used for heating appliances, hot water, cooking, and various other purposes also. LPG is also used as an alternate fuel in vehicles due to soaring in the prices of petrol and diesel.

Some people have low sense of smell, may or may not respond on low concentration of gas leakage. In such a case, gas leakage security systems become an essential and help to protect from gas leakage accidents. A number of research papers have been published on gas leakage security system [1-13]. Embedded system for Hazardous gas detection and Alerting has been proposed in literature [7]. Where the alarm is activate immediately, if the gas concentration exceeds normal level.

Bhopal gas tragedy was an example of gas leakage accident in India. This was world’s worst gas leakage industrial accident. Gas leakage detection is not only important but stopping leakage is equally essential. This paper provides a cost effective and highly accurate system, which not only detect gas leakage but also alert (Beep) and turn off main power and gas supplies, and send an SMS. GSM module is used which alert the user by sending an SMS [2]. In order to provide high accuracy gas sensor MQ-6 has been used.

II. METHODOLOGY USED

The functionality of system is divided into three main steps. The fig. 1 shows the block diagram of gas leakage security system.

Fig. 1 Block diagram of gas leakage security system.

In the initial step, the gas leakage is detected by the gas sensor MQ-6. This detects the gas leakage and gives the signal to the microcontroller with the help of ADC. After that in second step the microcontroller receive the signal, send by gas sensor. It sends activation signal to other external devices attached with it. Such as two stepper motor IC (ULN 2003A), buzzer, LCD (Liquid crystal display), GSM module [10] and RF link. In the last step, many tasks have been performed such as buzzer activates simultaneously message display on...
liquid crystal display screen, GSM module activated, which send warning SMS to the user. Stepper motor IC (ULN 2003A) to drives the stepper motor attached it, as a result main power and gas supplies turn off. At the end, when the gas leakage is successfully stopped then with the help of reset button the whole system reached to the initial stage.

A. MQ-6 Gas Sensor

MQ6 is a semiconductor type gas sensor which detects the gas leakage. The sensitive material of MQ-6 is tin dioxide (SnO₂). It has very low conductivity in clean air [4]. This Gas sensor not only has sensitivity to propane and butane but also to other natural gases, low sensitivity to cigarette smoke and alcohol. The MQ-6 gas sensor is shown in fig. 2. This sensor can also be used for detection of other combustible gas such as methane.

The concentration range of MQ-6 gas sensor is 300-1000 ppm. This sensor is available in 6 pins package, out of which 4 pins are used for fetching the signals and other 2 pins are used for providing heating current. This sensor has fast response time.

![Fig. 2 MQ-6 gas sensor [5]](image)

B. GSM Receiver

GSM module is used to send an SMS to the user cell phone [8]. When the gas leakage is detected by the gas sensor, microcontroller sends a signal to GSM module [2], in which one of the tasks is to send the text SMS. GSM module requires one SIM card [17]. This module is capable to accept any network SIM card. Fig. 4 shows a GSM module IC(Integrated circuit). This module has a unique identity number like mobile phones have. These module works on 12V DC supply [18]. We can send SMS and also send a voice message. These SMS or voice messages are saved in the microcontroller memory. Multiple SMSs can also be sends to user, police and fire station etc.

![Fig. 4 GSM module IC](image)

C. Stepper motor Driver

Two stepper motor has been used, both are connected to the stepper motor driver IC (ULN 2003A). A 12V external DC supply has been given to the stepper motor [9]. The main purpose of the stepper motor is to turn off the main power and Gas supply. One motor is used to turn off the main power supply. Motor is attached to a main switch in such a way that when a motor rotates 60°, then immediately power supply turn off. Now on the other hand, the second motor turns off the main gas supply. A mechanically coupled stepper motor to main gas knob, so that when motor rotates 180° then immediately the knob close.

The power need by the sensor is 5V. This sensor has different resistance value in different concentration. For an example, if we calibrate the MQ-6 gas sensor to the 1000ppm of propane concentration in air, then the resistance value would be approximately 20kΩ. The change in the resistance value with respect to the concentration as discussed above is shown in fig. 3.

![Fig. 3 shows the typical sensitivity characteristics of the MQ-6 gas sensor for several gases.](image)
D. Radio Frequency

RF transmission system composed of Amplitude Shift Keying (ASK) with the transmitter/receiver (Tx/Rx) pair, operating at frequency of 434 MHz. Transmitter modules takes serial input and transmits it through RF. Receiver module receives signals which are transmitted by transmitter module placed away from it. The RF module has been used with a set of four channels Encoder/Decoder ICs. HT12E & HT12D have been used as encoder and decoder ICs respectively. The encoder converts the parallel inputs into serial signals. These signals are serially transferred through RF. The decoders are used after the receiver to decode the signal and obtain the original signals as an output. These outputs can be easily observed on the corresponding LEDs. RF module is shown in fig. 5.

Fig. 5 RF module

III. RESULT

The prototype of the gas leakage security system has been shown in fig. 6. This system has been tested by taking a small amount of LPG gas near to the sensor. MQ-6 gas sensor detects the LPG gas and sends a signal to the microcontroller. After that microcontroller send an active signal to other externally connected devices. As a result a buzzer rings and a message is display on LCD screen. Simultaneously main power and gas supply turns off with the help of stepper motor and GSM module send an SMS [3]. When reset button is pressed, the system refreshes itself and whole system regains its initial position.

Fig. 6 Prototype model of proposed system

IV. CONCLUSION

In this system we have describe a new approach for gas leakage detection system at a low concentration. The leakage is detected with the help of MQ-6 gas sensor. Sensor sends a signal to microcontroller. In the next step microcontroller sends an active signal to other externally connected devices. The efficiency and memory of the microcontroller can be increased if Philips microcontroller is used in place of AT89C51.multiple SMS can be send by changing programming GSM module. To change the SIM card we have to make changes in program.

REFERENCES


Fusion of Reconstructed Magnitude and Phase part of MRI

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Abstract— The main objective of my research work is to fuse the denoised medical images, as the two parameters which plays important role in MRI (magnetic resonance imaging), are feature extraction and object recognition, which would be difficult if the images are corrupted with noise. Denoising is a challenging problem in MRI (acquired as a complex signal) as the magnitude and phase error both are to be corrected. Now for detailed diagnosis the MR Images can be taken more than once say one taken six months ago and one taken currently, so it is impossible to have the same alignment of the body part same as the previous scan so it would be difficult to compare MRI with naked eyes. So after getting denoised MRI, I am fusing the MR Images. I have developed an algorithm for MRI denoising in which I am using non-linear adaptive gradient smoothing filter for magnitude reconstruction and non-linear anisotropic diffusion for phase reconstruction and affine transformation for fusing the MR images linearly. I have got satisfactory results from my algorithm as applied on various live MR Images taken from the radiologist. Instead of using Gaussian filter or linear filter the non-linear gradient filter gives the better result as its smoothes the edges little hence preserving the fine details and borders. Anisotropic diffusion of the magnitude reconstructed MR Image gives sharp edges and better visualization than the simple FFT transform, hence phase part is reconstructed. Linear fusion using affine transform gives image with better features identified; hence better diagnosis and treatment can be done.

Keywords— MRI denoising, magnitude and phase reconstruction, non-linear gradient filter, affine transform.

I. INTRODUCTION

Medical images typically suffer from one or more of the following imperfections:
• Low resolution (in the spatial and spectral domains);
• High level of noise;
• Low contrast;
• Geometric deformations;
• Presence of imaging artifacts.

Either due to an instrument or human error it might be possible to have a noisy and blur image. So MR Image denoising is an important research field in medical [3,4,5,6]. It can be considered as a pre-processing step for other tasks such as image segmentation, image fusion and so on. While diagnosis of the human body two or more MR images needed to be taken sequentially or may be after some months after giving some treatment. It is quiet difficult to have the same alignment of the body during the consecutive scans and impossible if scanned after months. So for the accurate diagnosis images are needed to be compared or collage which is not possible to do with naked eyes or just by overlapping the images. So image fusion does the needful task and fused image give detail features helping for better diagnosis of disease and treatment can be given accordingly.

Flow chart of the proposed work is shown below in Fig. 1.

II. MR IMAGE DENOISING

I have studied the noise in MRI medical image modality which is having Rician distribution. Mainly low signal intensities (SNR < 2) are biased due to the noise. MRI being acquired has complex quantity has noise in real and imaginary part which can be differentiated by magnitude and phase part of the complex quantity. The magnitude and phase part of the MRI is shown below in fig. 2.
The noise observed in the MR Images is governed by Rician distribution. The probability distribution of a sinusoid plus narrow band noise is called the Rice Distribution [18].

The magnitude of the image can be found out by calculating the magnitude of each pixel having real and imaginary values.

\[
\text{Magnitude} = \sqrt{\text{Real}^2 + \text{Imaginary}^2}
\]

Let \( M \) denotes the intensity of image having noise inclusion and in absence of noise image intensity is \( A \), then the probability distribution is given by expression

\[
p_M(M) = \frac{M}{\sigma^2} e^{-\frac{M^2+\Delta^2}{2\sigma^2}} I_0\left(\frac{AM}{\sigma^2}\right) \quad \text{[1]}
\]

where \( I_0 \) is the modified zeroth order Bessel function and \( \sigma \) is the standard deviation of the Gaussian noise in the real and the imaginary images [18].

This expression is similar to the Rice density i.e. Rician distribution and is plotted in Fig. 3 for different values of the SNR \( A/\sigma \).

\[
\tan^{-1}\frac{\text{imaginary part}}{\text{real part}}
\]

The distribution of the phase noise \([8]\), is given by

\[
p_{\Delta \theta}(\Delta \theta) = \frac{1}{2\pi} e^{-\frac{A^2}{2\sigma^2}} \left[ 1 + A\Theta e^{2\pi\Delta \theta^2} \cdot 12\pi \right] -Ae^{2\pi\Delta \theta^2} \cdot x22 \ dx \quad \text{[2]}
\]

In image regions where there is only noise, \( A = 0 \), Eq. [2] reduces to

\[
p_{\Delta \theta}(\Delta \theta) = \begin{cases} 
\frac{1}{2\pi} & \text{if } -\pi < \Delta \theta < \pi \\
0 & \text{otherwise}
\end{cases} \quad \text{[3]}
\]
A. CONTRAST ADJUSTMENT

By using the simple Matlab commands imadjust and imcontra we can enhance the resolution and contrast of the MR Image reasonably. This is shown below in Fig. 5.

Contrast enhanced MR Image accurately detect all malignant lesion, vessels, arteries and veins.

B. MAGNITUDE CORRECTION

Smoothing with Gaussian filter loss the fine details and also edges are blurred [6]. Instead of that we have use non-linear smoothing in which each pixel is treated with varying intensity depending on its neighboring value. In general,

If \((x,y)\) is a part of an edge \(\rightarrow\) apply little smoothing
If not a part of an edge \(\rightarrow\) apply full smoothing.

This idea can be implemented by using a gradient function as given below.

\[
\text{grad}(I) = \begin{bmatrix} \frac{\partial}{\partial x} I \\ \frac{\partial}{\partial y} I \end{bmatrix} \quad \quad \quad \quad [4]
\]

The proposed scheme for applying gradient smoothing is as follows:

1. Find the gradient of the original noisy image \(I\), as given by eq. [4]
   \[
   I_x = \frac{\partial}{\partial x} I \quad \rightarrow \text{Horizontal gradient}
   \]
   \[
   I_y = \frac{\partial}{\partial y} I \quad \rightarrow \text{Vertical gradient}
   \]

2. Find again the gradient of horizontal and vertical gradient,
   \[
   [I_x, I_y] = \text{grad}(I_x)
   \]
   \[
   [\tilde{I}_x, \tilde{I}_y] = \text{grad}(I_y)
   \]

3. Find the gradient coefficient \(c\)
   \[
   c = \frac{1}{1 + \frac{1}{\sqrt{I_x^2 + I_y^2}}}
   \]

4. Calculate again \(\text{gradient}\) of horizontal and vertical gradient by multiplying with \(c\)
   \[
   [g_x, g_y] = \text{grad}(c \cdot I_x)
   \]
   \[
   [\tilde{I}_x, \tilde{I}_y] = \text{grad}(c \cdot I_y)
   \]

5. Calculate again \(\text{gradient}\) of horizontal and vertical gradient got from above procedure 4
   \[
   [g_{xx}, g_{yy}] = \text{grad}(g_x)
   \]
   \[
   [\tilde{I}_x, \tilde{I}_y] = \text{grad}(g_y)
   \]

6. Add the horizontal and vertical gradient of the procedure 5
   \[
   g = g_{xx} + g_{yy}
   \]
   
   denoised image \(\tilde{I}\) = noise image\(I\) – constant \(\times g\)

Apply the above procedure to the noisy image, the non linear smoothing gives good intraregion smoothing as
well as doesn’t do much with interregion smoothing [4, 6] (edges and lines) as shown in Fig. 6.

C. PHASE RECONSTRUCTION

For the phase reconstruction, I have proposed a scheme that uses an anisotropic method i.e. depends on the direction applied. For a gray scale image \( I \), the anisotropic diffusion is defined as

\[ \frac{\partial I}{\partial t} = \text{div} \left( \nabla (\mu \sum \nabla ) + \nabla (x, y, t) \right) \Delta I \] ........................ [5]  

\[ \frac{\partial I}{\partial t} = \Delta I = I_{xx} + I_{yy} \] ........................ [6]

where \( \Delta \) denotes the laplacian operator, \( \nabla \) denotes the gradient operator, and \( \text{div}(...) \) is the divergence operator and \( \nabla (x, y, t) \) is “edge stopping” function and is usually chosen as a function of the image gradient so as to preserve edges in the image [7,8,9,10,11].

Initial condition can be defined by putting \( t = 0 \) in equation [5].

\[ I(x, y, 0) = I_0(x, y) \]

Apply to image pixels

\[ J = (x_1,x_0) \times (y_1,y_0) \times [0,+] = D \times [0,+] \] ........................ [8]

Such as \( \frac{\partial J}{\partial n} = 0 \) in \( \partial D \times [0, +\infty] \),

where \( n \) is the normal vector at the image boundary.

![Original MR Image](image1)

(c) (d)

Fig. 7 (a) Original MR Image (b) zoomed part of the original MR Image.
Phase correction through (c) FFT (d) ANISOTROPIC DIFFUSION

Simply applying FFT also reconstructs the phase part of MR Image as shown in fig. 7 (c), as the Fourier analysis splits the real and imaginary part, can provide the significant information.

\[ F(x,y) = \sum_{m=0}^{\infty} \sum_{n=0}^{\infty} \tilde{f}(m,n) e^{-2\pi i (1) m x + n} \] ........................ [9]

But fine detail edges are not preserved, they are hazy. While applying anisotropic diffusion gives better result, as shown in fig. 7 (d) preserving the fine details and edges also.

III. LINEAR IMAGE FUSION USING AFFINE TRANSFORM

I have used affine transformation for the Fusion process to match the points in the two spaces. In an affine transformation for each point (considering suffix 1- X position, 2-Y position) \((a_1, a_2)\) in a MR Image a mapping can be defined into the coordinates of the another space \((b_1, b_2)\) such as

\[ b_1 = \mu_1 a_1 + \mu_2 a_2 + \mu_3 \]

\[ b_2 = \mu_4 a_1 + \mu_5 a_2 + \mu_6 \] ........................ [10]

where \( a_1, a_2 \) denote spatial coordinates in the source image and \( b_1, b_2 \) denote spatial coordinates in the target. Depending on the values \( \mu_i \), where \( i,j \) is from 1 to 3, certain well known geometric transformations can result[13].

The affine transformation preserves the straightness of lines, and hence, the planarity of surfaces and it preserves parallelism, but it allows angles between lines to change. It is an appropriate transformation class when the image may have been misaligned during acquisition [14, 15].

One way of defining \( \mu_i \) is by trigonometry terms, as follows

\[ \begin{bmatrix} b_1 \\ b_2 \end{bmatrix} = s \begin{bmatrix} \cos(\theta) & -\sin(\theta) & 0 \\ \sin(\theta) & \cos(\theta) & 0 \end{bmatrix} \begin{bmatrix} a_1 \\ a_2 \\ 1 \end{bmatrix} + \begin{bmatrix} \Delta x \\ \Delta y \end{bmatrix} \]

Unknowns parameter \( s \), \( \theta \), \( \Delta x \) and \( \Delta y \) can be calculated by taking four known coordinates \((x(a_1, a_2), x(y(b_1, b_2), y(b_1, b_2)), y(b_1, b_2), y(b_1, b_2))\), where \( x,y \) are the points of the source image and \( y,x \) are the points of the target image.

Solving the above equation in terms of \( b_1 \) and \( b_2 \), we get

\[ b_{11} = s \cdot (\cos(\theta) \cdot a_{11} - \sin(\theta) \cdot a_{21}) + \Delta x \]

\[ b_{21} = s \cdot (\sin(\theta) \cdot a_{11} + \cos(\theta) \cdot a_{21}) + \Delta y \]

\[ b_{12} = s \cdot (\cos(\theta) \cdot a_{12} - \sin(\theta) \cdot a_{22}) + \Delta x \]

\[ b_{22} = s \cdot (\sin(\theta) \cdot a_{12} + \cos(\theta) \cdot a_{22}) + \Delta y \]
Solving equation in terms of $s$, $\theta$, $\Delta x$, and $\Delta y$, we get

\[
\begin{align*}
    s &= \frac{b_{12} - b_{11}}{\cos(\theta) \cdot (a_{21} - a_{11}) - \sin(\theta) \cdot (a_{22} - a_{12})} \\
    \theta &= \tan^{-1}\left\{ \frac{(a_{12} - a_{11}) \cdot (b_{22} - b_{21}) - (b_{12} - b_{11}) \cdot (a_{22} - a_{21})}{(a_{22} - a_{21}) \cdot (b_{22} - b_{21}) - (b_{12} - b_{11}) \cdot (a_{12} - a_{11})} \right\} \\
    \Delta x &= b_{11} - s \cdot (\cos(\theta) \cdot a_{11} - \sin(\theta) \cdot a_{21}) \\
    \Delta y &= b_{21} - s \cdot (\sin(\theta) \cdot a_{11} + \cos(\theta) \cdot a_{21})
\end{align*}
\]

The algorithm for fusing the image is shown in fig. 8, step wise implementation.

IV. SIMULATION, EXPERIMENTAL RESULTS & ANALYSIS

The property of the affine transformation is used for the matching of the common points in the images. The algorithm is developed in the MATLAB and is efficient enough to reconstruct the magnitude and phase of the complex MR Image and detect the matching and unmatching points and efficiently fuse the set of images.

The Fig. 9 and 11 shows the matches that are identified between the set of MR Images to fuse. Fig. 10 and 12 shows the resultant Fuse image. Each and every corner points are identified giving better features identified, in fig 12 the blood clot which is increased is clearly seen in the Fused image.

Fig. 9: Matches identified between MR Images

Fig. 10: Resultant Fusion MR Image
The performance matrices for the noisy and denoised MR Images used are MSE and PSNR.

<table>
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<th>Image 1</th>
<th>Image 2</th>
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<tbody>
<tr>
<td>MSE</td>
<td>44.1721</td>
<td>37.0968</td>
</tr>
<tr>
<td>PSNR</td>
<td>31.6793</td>
<td>32.4374</td>
</tr>
</tbody>
</table>

I have applied this algorithm to various MR Images as well as different medical imaging modalities getting better results recommended by doctors and radiologists.

V. CONCLUSION

The Denoising algorithm is quiet efficient to reconstruct the magnitude and phase of the complex MR Image and other medical image modalities. Instead of using Gaussian filter or linear filter, the non-linear gradient filter gives the better result as it smoothes the edges little hence preserving the fine details and borders. Anisotropic diffusion of the magnitude corrected MR Image gives sharp edges and better visualization than the simple FFT transform, hence phase part is reconstructed. Affine transformation gives image with better features identified, hence giving the linearly fused medical images.

In future for comparison and finding of similarities of disease between two patients, same algorithm can be developed for diagnosis of intrapatient medical images. Also the Algorithm can be modified for 3D MR Images and other medical imaging modalities.

ACKNOWLEDGEMENT

I am grateful to my guide Prof. Ashish M. Kothari and to AITS, Rajkot, Gujarat, India.

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Comparative Analysis of Denoising Methods in CT Images

Tarandeep Chhabra, Geetika Dua and Tripti Malhotra

Abstract: In CT Scan there is a scope to adapt patient image quality and dose. Reduction in radiation dose (i.e. the amount of X-rays) affects the quality of image and is responsible for image noise in CT. However, several denoising algorithms can be used to improve the image quality. This paper contains the comparative analysis of a number of denoising algorithms namely median filtering, wavelet decomposition, wave atom decomposition, wiener filtering, anisotropic diffusion and NLM means filtering. We compute some quantitative performance metrics like PSNR, SNR, MSE, S/MSE and MAD. This comparison helps in the assessment of image quality and fidelity.

Keywords: computed tomography, noise reduction, filtering, denoising, SNR, MSE.

I. INTRODUCTION

CT Scan stands for Computed tomography. It basically uses X-rays to obtain structural and functional information about the human body. In CT, the image quality is influenced by many technical parameters. One of the most important parameter is the radiation dose. The quality of image increases with the significant amount of radiation dose [1]. But an increased amount of X-rays being absorbed by the human body increases the chances of cancer. So we need to reduce the radiation dose which is responsible for image noise in CT. So for proper analysis and diagnosis it is required to reduce the image noise and filtering is thus applied to clear such images. Any noise reduction algorithm aims to enhance the fidelity of an image which actually means removing the random and uncorrelated structures and retaining the resolution. Denoising of image data has been an active area of research and different methods such as median filtering, wiener filtering, use of NLM filter, wavelets decomposition, wave atom, bilateral filtering, isotropic, anisotropic diffusion, etc have been used.

II. METHODS TO REDUCE NOISE IN CT SCAN IMAGES

A. Median Filter

Linear filters are generally used to reduce noise in CT images. Here, the neighboring pixels represent the additional samples of the same value as that of reference pixel. In linear filters the convolution process is used for implementing the neighboring kernels as neighborhood function. But this may lead to the blurring of edges. To overcome such a problem non-linear filters are used for noise reduction. These filters help to preserve edges. Median filter is an example of non-linear filters. In median filter, the ranking of the neighboring pixels is done according to the intensity or brightness level and value of the pixel under evaluation is replaced by the median value of surrounding pixel values.

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Neighborhood values: 115, 119, 120, 123, 124, 125, 126, 127, 150
Median value= 124
Median filter can therefore effectively denoise medical images. The images distorted or blurred by shot or impulse noise can excellently denoised using this filter. Median filters have many advantages over smoothening filters[9]:

- In median filter the output values consist of only those present in the neighborhood (median value) so there is no reduction in contrast across the steps.
- The boundaries are also not shifted when median filter is used.
- The edges are minimum degraded and hence median filter can be repeatedly applied.

B. Wavelet Decomposition

The term Wavelet means a short wave like oscillation. Its amplitude value starts from zero, increases, again degrades to give zero value.
Figure 1: Wavelet

Wavelet transform is actually a mathematical technique which is used to synthesize a signal in time domain. It can be combined with any unknown signal to analyze that signal[8]. Then the transform of each segment is computed. At high frequencies this transform gives poor frequency but good time resolution, and vice-versa.

\[ s(i) = \sum_k c_{jk} \phi_j(k) + \sum_k \sum_l w_{jk} \psi_{jl}(k) \]

where, \( \psi_{jl}(x) = 2^{-j/2} \psi(2^j x - l) \) and \( \phi_j(x) = 2^{-j/2} \phi(2^j x - l) \).

Threshold is applied to the wavelet coefficients for noise reduction. Threshold is further of two types:

(i) Soft threshold
(ii) Hard threshold

Soft and hard threshold can be given as:

\[ \rho_{hard}(x) = \begin{cases} x, & |x| < \lambda \\ 0, & \text{otherwise} \end{cases} \]

\[ \rho_{soft}(x) = \begin{cases} x - \lambda, & x \geq \lambda \\ x + \lambda, & x \leq \lambda \\ 0, & \text{otherwise} \end{cases} \]

Further, there are a number of basis functions that can be used as mother wavelet for wavelet transformation. Commonly used wavelet functions are haar, daubechies, coiflet, symmlet, etc. the wavelets are chosen based on their shape and their ability to analyze the signal in a particular application.

C. Wave atom Transform

Wave atoms are used for harmonic computational analysis. These are a variant of 2D wavelet packets that retains an isotropic aspect ratio [6]. They have a sharp frequency localization that cannot be achieved using a filter bank based on wavelet packets. Wave atoms obey the parabolic scaling law: wavelength \( (diameter)^2 \). The name “wave atom” comes from the property that they provide an optimally spars representation of wave propagator with applications to fast numerical solvers for wave equation.

D. Wiener Filter

Wiener filters are basically the optimum linear filters which involves linear estimation of a desired signal sequence from another related sequence while solving the linear filtering problem certain important parameters like mean and correlation functions of useful signal and unwanted additive noise are assumed. Now the aim is to design a linear filter whose input is the noisy data and output is required to minimize the effect of noise[10]. The filter optimization problem is to minimize the value of MSE that is defined as mean square value of error. It is the difference between the desired and the actual filter output. The resulting solution (for stationary inputs) is called the wiener filter. However, wiener filter is inadequate for non stationary inputs because in such a case the optimum filter has to assume a time varying form.

In wiener filter, the performance function is given as:

\[ x = E[|e(n)|^2] \]

This is called “mean square error criterion”

E. Anisotropic Diffusion

Diffusion filtering of an image is similar to the physical diffusion process which provides equilibrium while following the law of conservation of mass. The image intensity can be seen as ‘concentration’. The noise can be seen as noise inhomogeneities. The inhomogeneities can be smoothened by ‘diffusion’. Diffusion filtering is digital image processing is mainly of two types: linear and non linear. Since the diffusion process relates a concentration gradient with a flux, linear isotropic diffusion is that in which these quantities are parallel[2]. Linear isotropic diffusion is used for image smoothening. The main disadvantage is that it blur the important features like edges. In non linear diffusion the concentration gradient and flux are not parallel and the filter coefficients change in response to differential structures within the image[3].

The first inhomogeneous diffusion model (anisotropic diffusion) was given by Perona and Malik[7]. Their idea was to vary the noise removal in nearly homogeneous regions while avoiding any alteration of the signal along significant discontinuities. The discontinuities are edges in images that arise due to the sharp changes in image intensity. The change in intensity \( I \) over time was defined as[7]:
$I_x = \text{div}(g \| V I \| V I)$ with: $g(x) = \frac{1}{1 + e^{x \alpha}}$  \hspace{1cm} (5)

where $g$ is the conduction function.

**F. NLM Filter**

The aim of any denoising method is to recover the original image from a noisy environment,  
$v(i) = u(i) + n(i)$ \hspace{1cm} (6)  
where $v(i)$ is the observed value, $u(i)$ is the actual or the true value and $n(i)$ is the noise perturbation at a pixel $i$.

Several methods can be used to denoise and recover the true image $u$. One such method is to use NLM (non-local means) filter[4]. The NLM means algorithm is defined by the formula:

$$NL[u](x) = \int_{\Omega} e^{-\frac{1}{2n^2}[G_{\Omega}(x+\frac{i}{n}-u(y+\frac{j}{n}))^2]} u(y)dy$$  \hspace{1cm} (7)

where $x \in \Omega$,  
$C(x) = \int_{\Omega} e^{-\frac{1}{2n^2}[G_{\Omega}(x+\frac{i}{n}-u(y+\frac{j}{n}))^2]} dy$ is normalizing constant,  
$G_{\Omega}$ is the Gaussian kernel and $h$ acts as a filtering parameter.

According to this formula the denoised value at $x$ is the mean of all the values at all the points whose Gaussian neighborhood is as the neighborhood of $x$.

**III. COMPARISON PARAMETERS**

The final step is to compute comparison parameters to compare the results of all above described techniques.

**A. Mean Square Error (MSE)**

Mean square error is a dominant quantitative performance metric in the field of image processing. It is used for the assessment of image quality and fidelity. The cumulative squared error that occurs between compressed and original form of image is termed as MSE. It is mathematically defined as:

$$MSE = \frac{1}{m \times n} \sum_{i=1}^{m} \sum_{j=1}^{n} (N(i,j) - DN(i,j))^2$$  \hspace{1cm} (8)

where $m$ is the number of rows in the image, $N(i,j)$ is noisy image and $DN(i,j)$ is denoised image.

**B. Peak Signal To Noise Ratio**

PSNR is mathematically described as:

$$\text{PSNR} = 10 \log_{10} \left( \frac{R^2}{\text{MSE}} \right)$$ \hspace{1cm} (9)

where $R$ is the maximum fluctuation in the input image data type. For example, if the input image has a double precision data type, $R=1[6]$. The PSNR value approaches infinity as the MSE approaches zero. Higher value of PSNR represents higher image quality. Small value of PSNR represents high numerical differences between images.

**C. Signal To Noise Ratio**

The signal-to-noise ratio is a technical term used to characterize the quality of the signal detection of a measuring system. It is mathematically described as:

$$\text{SNR} = 10 \log_{10} \left( \frac{\text{var}(x)}{\text{var}(\hat{x}-x)} \right)$$ \hspace{1cm} (10)

where $x$ is the noise free simulated images and $\hat{x}$ is noisy or denoised image.

**IV. RESULTS AND DISCUSSION**

This paper contains the results, obtained after following the median filter, wiener filter, wavelet, wave atom, anisotropic and NLM denoising algorithms. Further, comparison parameters like PSNR, MSE, SNR, S/MSE are calculated and compared.
V. CONCLUSIONS

In this paper, we demonstrated the image denoising results obtained from various denoising algorithms namely median filtering, wavelet decomposition, waveatom decomposition, anisotropic diffusion, weiner filtering and NL-m filtering. By investigating the comparison parameters, it is clear that anisotropic diffusion technique is promising. This method smoothes the noise while preserving the important features like edges. This techniques provides the maximum SNR (i.e 0.7143), S/MSE (i.e 7.1733) and minimum MSE (i.e 0.060561). So, from the comparative analysis of all the above described denoising algorithms it is clear that anisotropic diffusion method is best among all discussed above.

Table 1: Comparison parameters

<table>
<thead>
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Design and Fabrication of Microstrip Antenna for UWB Applications

D.Punitharaj and S.Kalaimani

Abstract – In this paper, a new technique has been proposed for proving the performance of the micro strip antenna for enhancement of the bandwidth is obtained .the operating frequency proposed in this project is 3.1-10.6 GHz for UWB applications. The simulation is done by using HFSS software. The simulated results are compared using VSNA analyser. The simulated results in return loss, efficiency are compared with the ideal one. The main objective of the proposed work is to fabricate a micro strip antenna for ultra-wide band applications. The specifications of the designed antenna are 25 x 29 mm. It allows the frequency band of 3.1-10.6 GHz. The substrate used in this design is FR4. The thickness of the substrate is 1.6 mm. The efficiency of the antenna is increased by increasing the thickness. FR4 is mainly used because it has high amount of thickness compared to the other substrate. The main objective of the paper is to notch the narrow band applications which are interrupting in the ultra-wide band spectrum.

Index terms-Ultra wide band (UWB); micro strip patch antenna; radiation pattern; HFSS; VSNA analyser.

I INTRODUCTION

Ultra-Wideband (UWB) is a technology for transmitting information spread over a large bandwidth (>500 MHz)[1] that should, in theory and under the right circumstances, be able to share spectrum with other users. Regulatory settings of Federal Communications Commission (FCC) in United States are intended to provide an efficient use of scarce radio bandwidth while enabling both high data rate "personal area network" (PAN) wireless connectivity[2] and longer-range, low data rate applications as well as radar and imaging systems. Ultra Wideband was traditionally accepted as pulse radio[3], but the FCC and ITU-R now define UWB in terms of a transmission from an antenna for which the emitted signal bandwidth exceeds the lesser of 500 MHz or 20% of the centre frequency. Thus, pulse-based systems where in each transmitted pulse instantaneously occupies the UWB bandwidth, or an aggregation of at least 500 MHz worth of narrow band carriers, for example in orthogonal frequency-division multiplexing(OFDM)[4] fashion can gain access to the UWB spectrum under the rules. Pulse repetition rates may be either low or very high. Pulse-based UWB radars and imaging systems tend to use low repetition rates, typically in the range of 1 to 100 mega pulses per second. On the other hand, communications systems favour high repetition rates, typically in the range of 1 to 2 giga-pulses per second, thus enabling short-range gigabit-per-second communications systems[5].

Each pulse in a pulse-based UWB system occupies the entire UWB bandwidth, thus reaping the benefits of relative immunity to multipath fading (but not to inter symbol interference) [6], unlike carrier-based systems that are subject to both deep fades and inter symbol interference. The UWB characteristics are very well suited to short distance applications. A representative case is for PC peripherals; see wireless USB (implemented on top of UWB).

The outline of this paper is Section I. Introduction, Section II. Description about the antenna geometry, Section III. Results obtained from the antenna, Section IV. Conclusion

II ANTENNA GEOMETRY

The proposed work is a compact printed micro strip-fed monopole ultra wide band antenna with triple notched bands is presented and analyzed in detail. A straight, open ended quarter-wavelength slot is etched in the radiating patch to create the first notched band in 3.3–3.7 GHz for the WiMAX system. In addition, three semicircular half-wavelength slots are cut in the radiating patch to generate the second and third notched bands in 5.15–5.825 GHz for WLAN and 7.25–7.75 GHz for downlink of X-band satellite communication. Surface current distributions and transmission line models are used to analyze the effect of these slots. The antenna is successfully fabricated and measured, showing broadband matched impedance and good omnidirectional radiation pattern. The designed antenna has a compact size of 25 x 29 mm. There is an issue of a possible electromagnetic interference, as over the allocated wide bandwidth of the UWB system, some narrow bands for other communication systems exist, such as WiMAX operating in 3.3–3.7 GHz, WLAN operating in 5.15–5.825 GHz, and downlink of X-band satellite communication systems in 7.25–7.75 GHz. Three band stop filters connected to a UWB antenna can be used to reject these bands. However, this increases the complexity of the system. A simpler way to solve this problem is to design the UWB antenna with band-notched characteristics. UWB antennas with band-notched function have been reported, mostly with one notched band for WLAN in 5.15–5.825 GHz. Recently, several antennas with dual notched bands or triple notched bands were presented. In this letter, the detailed analysis and the operational principles of the embedded slots in getting the notched band function, both half-wavelength and quarter-wavelength types. A new term, an effective length of a slot, and use this concept along with the surface current distributions and transmission line models to analyze the physical effects of those slots.
III RESULTS AND DISCUSSIONS

In this research, a novel U-shaped antenna has been designed which provide an end fire radiation pattern when laid down in an array configuration. It has been observed that arrays consisting of 4 and eight elements produce an end fire radiation pattern when their current distribution is changed. The analysis procedure adopted for observing the change in radiation pattern is that initially the effect of change in current distribution on radiation pattern is analyzed for single U-shaped micro strip antenna. Afterwards, a two, four and eight element array have been designed, and the same analysis of change in radiation pattern with change in current distribution is observed. The critical aspect of designing an array is designing the array feeder. In the arrays designed, corporate feed networks with quarter wave transformers for impedance matching are used. SMA connector of 50Ω and FR-4 substrate is used. The extensive, rapid and explosive growth in wireless communication technology and communication systems is prompting the extensive use of low profile, low cost and easy to manufacture antennas. All these requirements are efficiently realized by micro strip antennas. Micro strip antennas grant RF engineers with innumerable advantages as compared to conventional antennas; such as small size, low profile, low cost, light weight, mechanically robust, easy integration in electronic and communication systems and bulk production.
IV CONCLUSION

The foremost objective of the proposed work is to design an antenna for UWB applications. The proposed antenna is designed to overcome the drawbacks of the existing antennas and to overcome the interference of narrow band in the ultra-wide band spectrum. The proposed antenna is used to notch the narrow band entering into the UWB. It can be used for short distance applications efficiently. The antenna has been highly reliable, good gain and has high percentage of accuracy. In this letter, a compact printed micro strip-fed triple band notched UWB antenna has been presented and analysed in detail. To obtain three notched bands, two types of slots a straight open-ended quarter-wavelength type and a semi-circular half wave length type were etched in the radiating patch. We introduced new term, an effective length of a slot, and used this concept along with the surface current distributions and transmission line models to analyse the physical effects of these slots generating the band-notched characteristics. The antenna was fabricated and measured, showing broad bandwidth, three designed notched bands, and good unidirectional radiation patterns. The substrate used in the proposed antenna is FR4. FR-4 (or FR4) is a grade designation assigned to glass-reinforced epoxy laminate sheets, tubes, rods and printed circuit boards (PCB). FR-4 is a composite material composed of woven fiber glass cloth with an epoxy resin binder that is flame resistant (self-extinguishing).

V FUTURE WORK

The proposed technique is designed mainly to overcome the interruption of narrow band in ultra-wide band spectrum. The proposed scheme is implemented using HFSS simulation software. The real time implementation of the project is majorly used in the short distances applications. The efficiency obtained from the proposed scheme is nearly 90%. The efficiency can be increased by changing the fractal cuts over the patch. By increasing the efficiency we can radiate the signal effectively.

VI REFERENCES


A Survey on Different Techniques used for Solving Job Shop Scheduling Problem

Deepa Chaudhary, Deepali Singh, Nancy Rawat, Prerit Agarwal, Piyush Kumar

Abstract: This paper aims to have a comparison of a few of those techniques which are already suggested by different researchers in order to find optimal solution for the problem of job shop scheduling. Many approaches such as different crossover operators, variation in mutation operators and constrained problem statement etc. have been applied in order to achieve this aim. Different techniques discussed here are critical block (CB) neighbourhood and disjunctive graph (DG) distance in crossover, fusion of crossover and local search, penalty function, penalty function with delay constraint and random keys for sequencing.

Keywords: comparison, genetic algorithm (GA), job shop scheduling problem (JSSP), optimization, survey.

I. INTRODUCTION

Job shop scheduling problem (JSSP) is not a very new problem being faced by industries. Since 1960’s researchers have been trying to unravel the mystery of JSSP and still are putting their efforts into it. The reason behind this problem being in research for so long is that one cannot be sure of finding schedule for m number of jobs occurring on n number of machines (with a processing time of their own) while making an optimal use of resources and time. That is why this has been considered as an N-P Complete problem where, we know that solution exists but one cannot assure the solution within polynomial time.

During the study, it has been found out that JSSP taken into consideration can be described as [1] [2] [3] [4] [5] [11] a problem existing on shop floor with m number of tasks on n number of machines.

- Here jobs are considered as activities that take place and machines are considered as resources that are to be used optimally.
- Technique constraint is applied on JSSP which says that an operation must be processed only after all its precedent operations are finished.
- Also, resource constraint is applied on JSSP which says that each job must be processed on each machine exactly once and one at a time.
- Time taken by first job processing on first machine till finishing last job processing on last machine is known as make-span of schedule. Researchers aim to minimise this make-span.
- Jobs must not be interrupted in between.

JSSP is such a complex combinatorial problem that instead of finding an exact solution, researchers look for an optimal solution in reasonable time by use of heuristics techniques. Different approaches have been proposed for solving this problem like branch and bound [6], dynamic programming [7], simulated annealing [8], priority rules, Tabu search [9], genetic algorithm [10].

Among many suggested techniques, genetic algorithms (GA) have been taken frequently into consideration for solving this problem.

Organization of rest of the paper is done in the following manner: section 2 gives a very brief overview of Genetic Algorithm, in section 3 different techniques used for solving a JSSP namely, critical block (CB) neighbourhood and disjunctive graph (DG) distance in crossover of GA, the penalty method for constraints in JSSP, fusion of crossover and local search, penalty function with delay constraints, random keys for sequencing and optimization respectively, have been discussed, section 4 presents a tabular comparison of these algorithms with a conclusion in section 5.

II. GENETIC ALGORITHM

Genetic algorithm (GA) was proposed by John Holland [10]. It follows the basic principle of Darwin’s theory of evolution. This principle states that only the strongest of all individuals survives over generations. GA repeatedly uses information present in solution population to generate new solutions with better performances.

GA involves few basic steps [11]:
1. Population initialisation
2. Fitness evaluation
3. Selection
4. Crossovers
5. Mutation
6. Termination criterion

III. DIFFERENT JSSP SOLVING TECHNIQUES USED IN DIFFERENT RESEARCHES

A. CB Neighbourhood and DG Distance in Crossover of GA [1]

A concept of critical block (CB) neighbourhood and disjunctive graph (DG) distance is used during the process of crossover in genetic algorithm.

For crossover based on CB neighbourhood and DG distance [1]:答辩
Considering two parents \( m1 \) and \( m2 \).
- Set \( z = z_i \), which generates CB neighbourhood for \( z, N(z) \).
- do
  - for each \( z_i \) in \( N(z) \), calculate distance with respect to \( m2 \) to produce \( D(z_i,m2) \).
  - sort \( D(z_i,m2) \) in ascending order.
  - starting from first in sorted \( D(z_i,m2) \) where \( z_i \) with probability=1
    - if fitness value is less than current fitness value i.e.,
      \[ \nu(z_i) \leq \nu(z) \]
      else accepted if probability=0.5.
- Starting from \( m1 \), modify \( z \) step by step towards \( m2 \).
- After sometime, \( z \) loses \( m1 \)'s characteristic and inherit \( m2 \)'s characteristic.
- Choose child depending on less DG distance between child and both its parents.

Here, in crossover and mutation, researchers used critical block neighborhood and the distance measured helped them to evaluate the schedules. Result has shown that the implementation of critical block neighborhood and the distance measure can lead us to the same result obtained by other methods.


Here, another new method of employing penalty on the solution, if the solution violates any constraint, was analyzed during the study. The penalty function can be explained through given equation:

\[
P(z) = n \cdot \alpha \cdot \left( \frac{\text{number of violations}}{n} \right) \cdot \sum_{i=1}^{n} [\alpha \cdot (z_i - t_i)]
\]

where,
- \( n \) is the number of jobs.
- \( ti \) is the due date.
- \( zi \) is the process time for job \( i \).
- \( n \) is the number of solutions.
- \( k \) is the iteration number.
- \( \alpha \) and \( \beta \) are constants.

The objective is to minimize the make-span, i.e., the final completion time of all the jobs. For a permutation, we can get a complete scheduling by decoding process.

C. Fusion of Crossover and Local Search [3]

Multistep Crossover Fusion (MSXF) is a new crossover operator in which local search functionality is built-in. A local neighbourhood search algorithm is used for base algorithm of MSXF.

Algorithm of MSXF [3]:
- We have two parents \( m1 \) and \( m2 \).
- Now we set \( x = m1 \) and \( z \).
- do
  - for each \( y_i \) that belongs to \( P(x) \), we calculate \( d(y_i, m2) \).
  - sort these \( y_i \) in ascending order of \( d(y_i, m2) \).
- do
  - randomly choose \( y_i \) from \( P(x) \) but giving more preference to the smaller value of \( i \).
  - calculate \( V(y_i) \) if \( y_i \) has not yet been visited.
  - if \( V(y_i) \leq V(x) \) then accept \( y_i \) with probability=1, else accept \( y_i \) with probability=\( P_i \).
- now, the index of \( y_i \) is changed to \( n \) from \( i \) and induces of \( y_i \) in \( [k+1, k+2, \ldots, n] \) from \( k \) to \( k-1 \)
- until \( y_i \) is accepted.
- set \( x = y_i \).
- set \( z = x \) if \( V(y_i) < V(z) \).

until some termination criteria is satisfied.

next generation uses \( z \).

D. Penalty Functions with Delay Constraints [4]

This method makes use of a penalty function as described in [2] with an addition of delay constraint. It has been analyzed that the problem considered was for 4 machines and 6 jobs where each job has 2 numbers of identical pieces. The delays of any kind, during the manufacturing of jobs at manufacturing site, are also considered during the algorithm generation. Hence, the inputs are processing time and delay time. Researchers have taken the processing time from a manufacturing unit in Northern India.

Assumptions made were [4]:
- The load/unload station capacity is unlimited.
- Each machine completely manufactures the job assigned to it.
- The jobs are atomic.

The inputs once set cannot be changed during the generation of the particular schedule.

E. Random Keys for Sequencing and Optimization [5]

Random keys can be defined as a method of solution representation using which many feasible offspring can be produced for various problems of optimization and sequencing. These random keys are used to represent a solution with random numbers. For decoding the solution random values are used as sort keys. Offspring feasibility problem can be eliminated by using chromosomal encoding.

Structure of random keys can be defined as [5]:
- Chromosomes are formed by generating random number for every modeling issue.
- A derived solution can be reached from these sorted random keys and prioritizing the keys from the order deduced.
• Instead of derived solutions, crossover operator is applied on random keys.

Random keys are considered so important because as they form feasible offspring solution after crossover.

Considering, an n jobs and m machine problem, generate an integer (1…m) randomly for each job and a uniform deviate (0, 1) is added to it. Here, machines are assigned according to the integer part of the random key and sequence is sorted according to fractional part. Assuming that jobs are processed at their earliest possible time, a schedule can be constructed.

IV. A COMPARATIVE STUDY

A tabular comparison between all the algorithms stated in the previous section (DIFFERENT JSSP SOLVING TECHNIQUES USED IN DIFFERENT RESEARCHES) is depicted in the table (TABLE 1).

TABLE 1 COMPARISON OF VARIOUS JSSP SOLVING GENETIC ALGORITHMS

<table>
<thead>
<tr>
<th>GA</th>
<th>Technique Employed</th>
<th>Chromosome Representation</th>
<th>Initial Population</th>
<th>Crossover Operation</th>
<th>Mutation Operation</th>
<th>Termination Criteria</th>
</tr>
</thead>
<tbody>
<tr>
<td>CB Neighbourhood &amp; DG distances in crossover of GA[1]</td>
<td>Critical block neighbourhood &amp; disjunctive graph distances</td>
<td>Job sequence matrix</td>
<td>Randomly generated</td>
<td>CB neighbourhoo d &amp; DG distances</td>
<td>If DG distance between parents is less than predefined value</td>
<td>Population size, number of iterations for crossover and mutation.</td>
</tr>
<tr>
<td>Penalty function for constraints in JSSP[2]</td>
<td>Penalty function</td>
<td>3-D vector &lt;j,m,d&gt;</td>
<td>Giffler &amp; Thompson algorithm</td>
<td>Repeated appending to parent 2 after drawing from parent1</td>
<td>Inverse mutation, Interchange mutation &amp; Insert mutation</td>
<td>Best solution obtained or total number of generations set.</td>
</tr>
<tr>
<td>Fusion of crossover and local search[3]</td>
<td>Multi step crossover fusion</td>
<td>A set of nodes with 0 as (start) &amp; * as (end) nodes</td>
<td>Equal number of left and right active schedules</td>
<td>Local search</td>
<td>Multistep Mutation Fusion when distance between parents is too small</td>
<td>Number of iterations.</td>
</tr>
<tr>
<td>Delay constraint with penalty function[4]</td>
<td>Delay constraint</td>
<td></td>
<td>Randomly generated</td>
<td>One point crossover</td>
<td>Transposition with 0.5 probability</td>
<td>Number of iterations.</td>
</tr>
</tbody>
</table>

V. CONCLUSION

By the comparisons done so far, we have realized that there is no specific algorithm for the computation of an optimal solution of JSSP. It is deduced after the comparisons that if we try to solve the problem with some constraints then a near
REFERENCES


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FPGA Implementation of a Bio inspired olfactory system for odor Identification and Classification using ANN Algorithm

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Abstract— This paper presents Electronic nose system is an artificial neural network system used to detect or classify odour of a specimen and it finds wide application in all commercial industries. To identify a new sample and then to estimate its concentration, use to both spike timing dependent plasticity learning techniques and the least square regression principle. In the first one is aimed at teaching the system how to discriminate among different gases, while the second one uses the least squares regression approach to predict the concentration of each type of samples. During the experiment, the odor data are sampled by a commercial electronic nose and are normalized before training and testing to ensure that the classification result is only caused by learning. The SNN has either a high or low output response for a given input odor, making it easy to determine whether the circuit has made the correct decision. The system aims at reducing the area overhead by incorporating a transposable SRAM array that share learning circuits which grows with the number of neurons also the system is trained for usage in chemical industry by coupling a chemo sensor array. All the component subsystem implemented on neuromorphic chip has been successfully tested in FPGA.

Keywords—Electronic Nose, Field programmable gate array, neuromorphic circuits, olfaction, Spike time dependent plasticity

I. INTRODUCTION

Currently the biggest market for Electronic Nose (EN) system used for all type of industries including Agriculture, Biomedical, Chemicals, Cosmetics, Environmental, Food, Manufacturing, Military and various research fields. In the electronic Nose system learning and classification algorithms plays an important roles [4].

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Nowadays, electronic nose are used for quality control application in the food industry, chemical industry and cosmetics industries and also improve the product purity. Currently it includes detection of odors specific to diseases for medical diagnosis and detection of pollutions and gas leakage for environmental protection. Electronic nose was developed in order to mimic human olfaction that functions as a non separative mechanism. That is odor and flavor preserved as a global finger print.

Spike time based training algorithm calculates the weight updates based on the temporal distribution of the pre synaptic spike or post synaptic spikes. This phenomenon is known as spike timing dependent plasticity (STDP) [1]. During the training, weight of the synapse is modified according to the timing difference between the pre-synaptic spike and the post-synaptic spike. When the post-synaptic spike occurs immediately after the pre-synaptic spike, the synaptic weight is increased. This type of synaptic is called the long-term potentiation (LTP). when, the synaptic weight is decreased, and this synaptic is called the long-term depression (LTD).

Recently the synaptic weight plasticity depends on not only the action potential timing but also on the post-synaptic depolarization. The LTP dominates at high firing frequencies. This mechanism can be used for two steady states (potentiated or depressed) and is widely implemented in the spike-based neuromorphic learning system. This work provides the integrated sensor technology and systems using spike time based neuromorphic models to implement an olfactory system in FPGA. The use of FPGA allows for low power operation, and reduces the silicon area. Most importantly, a neuromorphic Implementation detect the complex odor, in particularly odor segmentation and odor object identification in varying chemical environments to be solved with classical approaches Thus, the system has possible convenient sensing applications there is a requirement for the bias against a wide variation in background of odor signals.

In an olfactory bulb, the onset latency of the action potential decreases but the inter spike intervals remain constant as the odor concentration increases. As a result, onset latency could be a better way of representing odor concentrations in olfactory systems. The use of this characteristic determines the latency of a spike. A possible solution may be adding into the network a sub threshold oscillation, a phenomenon measured in biological neurons in which the voltage of the soma oscillates even without input stimuli. The latency can be defined as the timing difference between the start of the oscillation period and the first spike. In this work, the oscillation is not only treated as the time reference but also as an important characteristic for improving the classification performance.

In this paper, we present the Electronic nose system used to classify the different sample of odors and its represent the concentrations. The proposed spiking neural network takes advantages of sub threshold oscillation and onset latency.
representation to reduce power consumption and chip area, providing a more distinct output for each odour input. The rest of this paper contained as follows: In Section II, introduces the circuit blocks. Section III presents the results and discussion. Finally, Section IV offers the conclusion.

II. CIRCUIT IMPLEMENTATION

Fig. 1. Shows the system diagram. Olfactory network provides scalable olfactory system interconnecting the multiple chips. The olfactory sensors implemented in a resistive chemo sensor array employing carbon black (CB) sensing materials with signal processing circuitry. The sensor interface circuitry includes a dc cancellation circuit to sense the chemical odor molecules range. A spiking neural network forms the signal processing stage of the olfactory model. Learning circuit is dynamically adapting the weights for odor detection and classification. Learning is crucial for the design of an odor sensing system.

A. Receptors

Olfactory receptor neurons represent the front end of the mammalian olfactory system. Receptor cells respond to odorant molecules, and send the signal in to the glomeruli for preprocessing and encoding. Experimental data indicates that each glomerrulus receives axon from receptor cells. Mainly synapse onto the dendrites, act as a principal neurons (PNs). PN sends dendrite to a single glomerulus. Inhibitory neurons of the olfactory bulb make many principle neurons, forming a complex network represent the first stage of olfactory information processing. The output of PNs performs further processing.

B. Chemo sensor Array

The chemo sensor array consists of different sensor types respond to different chemical compounds. Such a various array potential has to increase the selectivity in the olfactory pattern recognition task while mimicking the function of the mammalian olfactory system. Chemosensory are transformed into spike trains by olfactory receptor models. These spike trains directly drive synaptic currents and summed for the purposes of signal enhancement. This summed signal provides the excitatory drive to PNs, provides the main output of the system. Lateral inhibitory neurons are used to both the output characteristics of the system. The sensor signals are fed in to forward through neural elements to principle neurons the network forms a distinct modular structure, of the glomerular organization for biological olfactory bulb model. We are implementing, each sensor cell contains a programmable current source, a sensor, and neuromorphic circuit. Each sensor has a dedicated set of circuitry, each of them biased and amplified optimally.

C. Signal Conditioning Circuitry
The neuromorphic network receives sensory signals from an array of chemo sensors which transform the molecular chemical information of an odorant into electrical signals suitable for processing in analog circuitry. The large variation of baseline signals for a chemo sensor array are caused by the poisoning effect during post-processing and different optimal operating.

An adaptive neuromorphic olfaction device consisting of on-chip chemo sensor array, on-chip sensors interface circuitry and neuromorphic olfactory current specifications for different sensor types. The large variation in baseline dc signals among the sensors may result in saturation of the subsequent signal conditioning amplifier stages thereby most important is loss of measurement range. Baseline signals of all sensors in the array are digitally stored using analog to digital converter. The output from the digital to analog converter provides the initial analog offset signal which is canceled using a difference amplifier. The output sensor interface circuit is maintained in analog continuous time domain and feeds into the subsequent neuromorphic circuit stage implemented on chip.

D. Synapse

Most of the researchers have proposed different circuits to implement the function of the synapses. The circuits integrate pre-synaptic spikes to a voltage and produce the EPSC according to this voltage. Due to the large number of synapses, occupied by the capacitor is very large. Biologically realistic neuron model is used, the input of the neuromorphic chip is provided by the non-biological sensor when the neuromorphic chip is integrated into an E-Nose. When the firing rate of the pre-synaptic neuron is constant and large enough, and the integrated current in the synapses is equal to the leakage current. Because the output of the pre synaptic neuron, adjust the current according to the synaptic weight. This modification reduces the circuit complexity, the chip area, and the power consumption.

To change the synaptic weight, the pre-synaptic spike and the post-synaptic spike should occur within a specific period, which is called the STDP window. This paper changes the STDP window according to the firing rate of the post-synaptic neuron to achieve a similar performance in learning with an improved training speed. The weight updates frequently (minimal frequency, 100 Hz) in the training phase, and the resolution of the weight is extremely low. Consequently, no refresh is required during training.

II. RESULT AND DISCUSSION

A. Chip information

circuits with on-chip STDP learning has been fabricated on a single chip using CMOS technology. Scalable olfactory system can be constructed by interconnecting multiple chips. Each sensor cell has an associated sensor interface circuit for dc cancellation, amplification and filtering. The outputs of sensor interface circuits feed to the inputs of the neuromorphic circuits. The sensor response and performance of the interface circuitry were characterized by delivering target vapors in the air. In the chip contained 3 spiking neural network arrays with each array having receptor neurons, synapses and principle neuron. The purpose of this chip was to test the circuit blocks implementing the neuromorphic systems. The learning in this chip was performed by off-chip.

B. Performance of the Olfactory system

The odor data were sampled by a commercial E-Nose product, the resistance of each sensor results was saved in the computer. The device samples the resistance of the sensor every 0.5–0.6 s. The device resamples the resistance baseline of the sensor when the gas experiments are performed to reduce the effect of background odor. The E-Nose ends a gas experiment by inhaling clean air (background air) for a short period to clean the sensors. The odor data were pre-processed and normalized by a self-written Lab View program and categorized into a training part and a testing part. The training part is fed into the neuromorphic chip by FPGA kit to perform the training. After the training procedure is completed, the testing part is fed into the chip and the classification result is shown by the computer.

In the electronic nose system are using chemo gas sensors. When using this type of sensor provides meaningful information and it’s calculated by the percentage of the resistance change before and after the sensor interacts with the odor molecule. When the reaction of the sensor to a particular odor is stronger, the resistance change is larger. The data have been preprocessed and normalized. The input voltage ranges from 0.35 V to about 1 V. The data need to be converted into threshold voltage format before fed into the chip. To make certain classification result is only caused by learning; the data need to be normalized before training and testing by fixing the total strength of the input stimuli. The self-written Lab View program extracts the percentage of the resistance change of each sensor, converting the data from a percentage change to the threshold voltage and normalizing it. The normalization procedure is as follows.

The information from the receptors is expressed as R1 after ~ Rn after and R1 before ~ Rn before. Rn after represents the resistance of the nth sensor after responding to the odor, and subscripted “before” represents the resistance before responding to the odor. The glomeruli implemented in the computer perform the following functions. Given n dimension data x which represents the percentage of reaction

\[
x(\mathbf{x}) = [x_1 \ldots x_n] + \mathbf{n} \rightarrow x_{n_{fix}} = \frac{x_{n_{invert}}}{\sum x_{n_{invert}}} \quad \frac{R_{1_{after}} - R_{1_{before}}}{R_{1_{before}}}
\]

Step 1 (Inverting Data): The threshold voltage is smaller when the sensor response is larger x_{n_{invert}} = \max (x_n) - x_n.

Step 2 (Fixing Total Stimulation Strength): When the data in different categories have levels of stimulation strength, the output of the neuromorphic network varies, even though the synaptic weights are all equal to a constant.

x_{n_{fix}} = x_{n_{invert}} / \sum x_{n_{invert}}
Step 3 (Adding Offset): After the second step, the minimal value in $x_n$ is zero. When the threshold is set to zero, the soma voltage of this neuron is never lower than the threshold level. 

$$x_{ni_{\text{new}}} = x_{ni_{\text{fix}}} + x_0.$$  

$x_{n_{\text{new}}} = [x_{1_{\text{new}}}, \ldots, x_{n_{\text{new}}}]$ is the actual input to the neuromorphic chip.

![Fig. 2. Mean firing rate of Neuron A and Neuron B](image)

Mean firing rate of each odor sample can be represented in the graph. For Neuron A odor sample of Ethanol although, the firing rate of neuron B is varied when different samples come in, It is larger than the firing rate of neuron A in most cases. In this experiment, the geometric mean is used to measure the total classification performance from each testing odor set.

### C. Neuromorphic Circuits Results

The neuromorphic circuits are presented in this section. A synapse circuit is tested by exciting spike input. The response for the synapse programmed with a weight input and time constant. Since the weight is negative with respect to the baseline, the synapse exhibits an inhibitory response. The output response of the on-chip STDP circuit over a two different pre synaptic and postsynaptic input spike. Initially, the synaptic weight is negative causing the synapse to generate an inhibitory response. The weight increment (decrement) is smaller if the pre synaptic spike follows the postsynaptic spike by a larger time interval. When a receptor neuron is connected to PN through a synapse. The input of receptor neuron firings at a rate of approximately 25 ms, the repeated receptor neuron firing and synapsing on to the PN eventually causes the PN to fire. The summation of exponentially decaying synaptic currents for multiple receptor neuron spikes. Importantly, this test result demonstrates the functional signal path through the entire neuromorphic circuit. The performance of the neuromorphic olfaction chip components are summarized in Table I.

![Fig 3. Finger print graph of Ethanol](image)

![Fig 4. ANN Simulation output](image)

![Fig 5. Area Usage](image)

Table I.
PERFORMANCE SUMMARY OF THE ADAPTIVE NEUROMORPHIC OLFACTION CHIP

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Values</th>
</tr>
</thead>
<tbody>
<tr>
<td>Technology</td>
<td>0.18µm</td>
</tr>
<tr>
<td>Supply voltage</td>
<td>5v</td>
</tr>
<tr>
<td>Chip area</td>
<td>1.78mm²</td>
</tr>
<tr>
<td>Sensor resistance</td>
<td>10kΩ to 200kΩ</td>
</tr>
<tr>
<td>Sensor driving current</td>
<td>1µA, 10µA, 100µA</td>
</tr>
<tr>
<td>Sensor bandwidth</td>
<td>&lt;1Hz</td>
</tr>
<tr>
<td>Synaptic time constant</td>
<td>10 ms to 300 ms</td>
</tr>
<tr>
<td>Weight range</td>
<td>± 1V</td>
</tr>
<tr>
<td>Neuron spike width</td>
<td>10µs–1ms</td>
</tr>
<tr>
<td>Neuron refractory time period</td>
<td>10 ms – 300 ms</td>
</tr>
</tbody>
</table>

All receptor neurons have a time constant of 200ms. The threshold voltages of all the RN’s are set at 70% of the peak sensor signal amplitude. To initiate learning in the network, the threshold voltages of PNs are chosen such that they spike on the arrival of first few correlated RN spikes. The refractory period of the RNs and the PNs are 60 and 120 ms, respectively. The STDP window function is set at ± 50ms. The circuits of the individual building blocks used in this network have been validated in the chip results. The proposed neuromorphic chip consumes less power and this is suitable for electronic nose system to classify the odor recognition and classification.

IV. CONCLUSION

This paper proposed Field Programmable Gate Array based adaptive neuromorphic circuits were inspired by the mammalian olfactory system. In this neuromorphic chip composed of chemo sensor arrays and STDP synapses, utilizes sub threshold oscillation to represent the strength of the stimuli resulting in lower power consumption, smaller chip area, and more distinct output for each odor input. The neuromorphic chip has been fabricated by CMOS process. The chip can be trained to classify odor categories at the same time combining two or more neuromorphic chips, the classification task can be performed by the electronic nose system. For the testing odor data set, provides the better classification performance and accuracy. As a result, the proposed neuromorphic chip is suitable for application in an E-Nose system for odor recognition and classification.

REFERENCES

Novel Dual-Band CPW-fed Monopole Antenna for WLAN/WIMAX Applications

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Abstract: A novel dual band design of a finite ground coplanar waveguide (CPW)-fed monopole antenna is presented for satisfying wireless local area network (WLAN) and Worldwide Interoperability for Microwave Access (Wi-MAX) applications. The proposed antenna consist of U-shape planar patch element and has frequency bandwidth of 842 MHz (1.95 GHz-2.79 GHz) and 1.08 GHz (4.86 GHz-5.94 GHz) for WLAN and Wi-MAX applications respectively. The basic theory and design are analyzed, and simulation using CST Microwave Studio commercial software is employed to optimize the antenna properties.

Keyword: CST Microwave Studio, WLAN communication standards and Wi-MAX communication standards.

1. INTRODUCTION

Multiband printed monopole antennas have widespread applications, especially in low power wireless communication gadgets. In the era of modern wireless communication system, dual band or multiband antennas with omni-directional radiation characteristics plays a vital role [1, 2]. The IEEE standard was proposed in 1997 for WLAN applications. The bands for WLAN applications are 2.4 GHz (2.400 GHz-2.484 GHz), 5.2 GHz (5.150 GHz-5.350 GHz) and 5.8 GHz (5.725 GHz-5.825 GHz) and for Wi-MAX applications bands are 2.5 GHz (2.5GHz-2.69 GHz), 3.5 GHz (3.4 GHz-3.69 GHz) and 5.5 GHz (5.25 GHz-5.85 GHz). Wi-MAX technology [3] is the most rapidly growing area in modern wireless communication [4]. The planar monopole antenna has received much more interest than others, due to its potential in providing the various radiation features required for dual band or multiband, wide bandwidth, low profile communication system. However, these kinds of antennas mostly need a large ground plane, which is often printed on opposite side of substrate from the radiating plane. Recently coplanar waveguide (CPW)-fed monopole antenna has become very popular in WLAN and Wi-MAX systems, owing to its many attractive features such as, wider bandwidth, low radiation loss, simple structure of a single metallic layer and easy integration with WLAN integrated circuits [5].

In this paper, a proposed antenna design with CPW feed technology has been used to achieve dual band operations for WLAN and Wi-MAX bands. The proposed dual band antenna consist of U-shape patch element, capable of generating two separate bands with good impedance matching conditions. This way, the antenna can achieve dual band performance to simultaneously cover the most commonly used 2.4 GHz/5.2 GHz/5.8 GHz WLAN bands and 5.5 GHz Wi-MAX bands. Detail of the proposed antenna design are described in the paper, simulated results are presented and discussed in the following section.

2. ANTENNA DESIGN

Fig.1 shows the geometry of proposed finite ground coplanar waveguide (CPW) fed dual band monopole antenna. The proposed antenna is simulated on FR4 substrate with dielectric constant 4.4 and thickness 1.6 mm. The U-shape patch element is chosen with dimensions of width W, length L and with vertical spacing of ‘d’ away from the ground plane. A conventional CPW-fed line designed with a fixed signal strip thickness L2 and gap distance of ‘g’ between the signal strip and the coplanar ground plane is used for exciting the radiating patch element. Two finite ground planes with the same size of width Wg and length Lg, are situated symmetrically on each side of the CPW feeding line.
The final optimized dimensions of proposed antenna are:

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Specification</th>
</tr>
</thead>
<tbody>
<tr>
<td>L</td>
<td>25 mm</td>
</tr>
<tr>
<td>W</td>
<td>48.32 mm</td>
</tr>
<tr>
<td>L₁</td>
<td>2 mm</td>
</tr>
<tr>
<td>W₁</td>
<td>28.4 mm</td>
</tr>
<tr>
<td>W₂</td>
<td>3.7 mm</td>
</tr>
<tr>
<td>Wg</td>
<td>12.62 mm</td>
</tr>
<tr>
<td>g</td>
<td>1.095 mm</td>
</tr>
<tr>
<td>Lg</td>
<td>8.275 mm</td>
</tr>
<tr>
<td>L₂</td>
<td>6.26 mm</td>
</tr>
</tbody>
</table>

Thickness of substrate (h) = 1.524 mm
Substrate permittivity = 4.4

3. RESULTS AND DISCUSSION

The simulated return loss results are shown in Figure 2. The bandwidth of 842 MHz (1.95 GHz – 2.79 GHz) for WLAN / WiMax has been achieved resonating at 2.23 GHz with the corresponding value of return loss as -20.14 dB. The bandwidth of 1.08 GHz (4.86 GHz – 5.94 GHz) also has been achieved resonating at frequency 5.28 GHz with the corresponding value of return loss -54.63 dB. The antenna covers WLAN standard (2.4 GHz, 5.2 GHz and 5.8 GHz)/WiMax standard 5.5 GHz band.
Figure 2: Simulated Return Loss Curve

Figure 3: Curve showing antenna characteristic impedance

The achieved value of return loss is good enough and frequency is closed enough to the specified frequency band 5.5 GHz for WiMax application / 2.4 GHz, 5.2 GHz and 5.8 GHz for WLAN application. The CPW fed antenna consist of U shape patch element. As the length of parallel strip of U shape patch element increases, the return loss of 1st band increases but 2nd band shifts towards lower band and vice versa. As the width of feed line increases the return loss of 2nd band increases sharply.

The maximum achievable gain over the entire frequency band of 2.23 GHz and 5.28GHz is 2.50 dB and 4.74 dB respectively. The achieved antenna impedence is 50 ohm as shown in Figure 3, which is very equal to the required impedence of 50 ohm. Figure 4(a) shows the simulated 3-D radiation pattern showing directivity at 2.23 GHz. Fig. 4(b) shows the simulated radiation pattern showing directivity at 5.28 GHz.
It shows that the directivity of proposed antenna is 2.29 dBi at resonating frequency 2.23 GHz. It also shows that the directivity of proposed antenna is 4.68 dBi at resonating frequency 5.28 GHz.

4. CONCLUSION

A dual band CPW fed monopole antenna suitable for WLAN/WiMax application is proposed. Using U shape patch element, a resonant mode having excellent impedance performance is achieved. Various parameters like gain, directivity, bandwidth and return loss are also studied. The return loss value -20.14 dB and -54.63 dB suggest that there is a good impedance matching at frequency point below -10dB. An omni directional radiation pattern result has been obtained which seems to be adequate for the envisaged applications. However, the size of microstrip antenna, reported here, is not very small. The gain of antenna is small but it can be increased using gain enhancement techniques.

References


Analog to Digital Converter (ADC) Review

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Abstract: This paper presents basic introduction about analog to digital conversion process. This is done by two steps: Sampling and Quantization. During the conversion process some parameters are more important such like resolution, SNR, SFDR which are introduced in to this paper. The performance of the ADC is measured by INL, DNL and quantization error which are also defined. There are different types of ADCs which are also introduced using with some literature regarding to this analog to digital conversion.

Keywords: Analog to Digital converter (ADC), pipeline, sample and hold, Digital to Analog converter (DAC), CMOS.

I. INTRODUCTION

Generally the data available from real world are in analog form, but this analog data storage procedure is more complex compare with digital storage. There is also more advantage like digital data storage requires less memory compared to analog data, less noisy and provides good encryption & security features [4], [19]. Therefore we required the analog to digital converter (ADC). Hence ADC plays an essential role in all kinds of electronics systems. As we know digital signal processing (DSP) is more popular today in this case ADC is required because of requirement of digital data only. Using this feature it is very popular in very large scale integration (VLSI). Analog to digital converters are useful building blocks in many applications such as data storage read channel [1] and an optical receiver [2] because they represent the interface between the real world analog signal and the digital signal processors. Recently digital multimedia application is more popular in this case ADC is required. In LED & LCD displays digital data is also required [5].

II. HOW TO CONVERT AN ANALOG SIGNAL IN TO DIGITAL SIGNAL

ADC is a device that uses sampling to convert continuous time signals in to discrete digital form. [4] ADC is such as an electronic device that converts an input analog voltage or current to a digital number proportional to the magnitude of the voltage or current. [3], [21] This process is done by two steps: Sampling and Quantization. Digitizing the coordinate value is called sampling and digitizing the amplitude value is called quantization. Sampling does the breaking down the analog value in to set of discrete values & quantization takes only a few discrete values from the available value and converting this discrete value into the digital values through the encoding process.

[5] For example, we have 0-8V peak-to-peak signal. Separate them into discrete states with 2V range for each state, so that we have 4 different quantization levels as shown in to the table and corresponding assign the digital value for each quantization level.

<table>
<thead>
<tr>
<th>Voltage range(V)</th>
<th>Quantization levels</th>
<th>Encoding</th>
</tr>
</thead>
<tbody>
<tr>
<td>0 – 2</td>
<td>1</td>
<td>00</td>
</tr>
<tr>
<td>2 – 4</td>
<td>2</td>
<td>01</td>
</tr>
<tr>
<td>4 – 6</td>
<td>3</td>
<td>10</td>
</tr>
<tr>
<td>6 – 8</td>
<td>4</td>
<td>11</td>
</tr>
</tbody>
</table>

Sampling frequency for the conversion process as per the Nyquist rate is at least twice of the maximum frequency of the input analog signal for the faithful reproduction of the input signal. For example, input signal is [3], [21-22]

\[ X(t) = A \cos(2 \pi \times f_m \times t) \] ........................ (1)

Where, \( f_m \) is the input signal frequency, then the sampling frequency is \( f_s = 2f_m \) as per the Nyquist rate [19].

![Sampling and Quantization](image)

This figure shows 4-bit (16 level) ADC sampling a sine wave input analog signal in to digital signal. Digital data may have any value from its 16 level.

III. ADC PARAMETERS

There are several parameters are used to test the performance of the conversion process.

A. Resolution

The resolution of the converter indicates the number of discrete value it can produce over the range of analog values [6-8]. It is usually expressed in bits. [20] For example, ADC
with a resolution of 8-bits can encode an analog input to one in 256 different levels, because of \(2^8 = 256\). It ranges from 0 to 255 or -129 to 127. Resolution can also be defined electrically, and expressed in volts. The resolution \(Q\) of the ADC is equal to the least significant bit (LSB) voltage \([5]\),

\[
Q = \frac{V_{\text{high}} - V_{\text{low}}}{2^m - 1} \quad \text{............... (2)}
\]

Where, \(m = \text{resolution in bits}\)

[5] For example, \(X(t) = A \cos(t)\) where \(A = 5V\). It swings from -5 to 5 volts. If ADC resolution is 8 bits: \(2^8 - 1 = 255\), and hence \(Q = (10 - 0) / 255 = 39\text{mV}\). Resolution of converter is limited by the signal-to-noise ratio (SNR) that can be achieved for a digital signal.

**B. Signal to noise ratio (SNR)**

SNR is most important parameter of the ADC. It measures the signal power and noise power for the output digital signal and give the amount of signal is distorted \([3-4], [6]\). So that high SNR is acceptable.

**C. Total harmonic distortion (THD)**

During the conversion process some samples are mixed due to noise and generate distorted signal. It is caused by ADC non-linearity, which generates harmonics of the input signal; it is undesired at the output stages \([6], [25]\). So that THD gives amount of harmonics are generated in to digital signal. It is measured in db as shown in to the figure 2.

**D. Spurious free dynamic range (SFDR)**

Dynamic range is a range for which output follows to input as per the requirements. After a sometime harmonics are generated due to non-linearity, so that the operating ranges for which spurious response are minimal is called the spurious free dynamic range \([6], [25]\). SFDR is always given at a particular frequency as shown in figure 3. It is the difference between primary and next highest spur as shown in to the figure 3.

**E. Power dissipation**

It gives amount of power dissipated by a specific ADC. Less the ADC circuit area less the power consumption \([3], [22]\). So that fabricates the ADC by the CMOS technology to cover less area.

### IV. ADC ERRORS

There are several errors are generated during conversion process as like:

**A. Aliasing**

If input signal is changing much faster than the sampling rate, then the digital signal reconstruction of the analog signal is not same as original one, but some spurious signal is generated is called aliases. This problem is called aliasing. \([5]\)

For example, 2 KHz sine wave being sampled at 1.5 KHz would be reconstructed as a 500Hz sine wave. To avoid aliasing, an input signal to an ADC must be low-pass filtered to remove frequencies above half of the sampling rate. This filter is called anti-aliasing filter \([6]\). Aliasing error is shown in to the figure 4.
B. Quantization error

Quantization error (quantization noise) is the difference between the original signal and the digitized signal as shown in figure 5 [4], [5]. Hence the amplitude of the error at the sampling instant is between zero and half of one LSB ($\pm 1/2$ $LSB$) . It is due to the finite resolution of the digital signal and it is unavoidable in all types of ADCs. This error is intrinsic to any types of ADCs. It is reduced by higher resolution of ADC [6].

C. Non-linearity

All ADCs suffer from non-linearity errors caused by their physical imperfections, causing their output to deviate from a linear function of their input. These errors can be mitigated by calibration techniques or prevented by testing. [5] A Non-linearity error reduces the dynamic range of the signals, also reducing the effective resolution of the ADC. This error is intrinsic to any type of ADCs. The most important parameters for linearity are Integral non-linearity (INL) and Differential non-linearity (DNL).

I. Integral non-linearity (INL)

It is the maximum deviation between an actual code transition point and its corresponding ideal transition point. INL is measured in LSBs, and calculated after offset and gain errors have been compensated [6], [25]. A positive INL indicates transition occurring later than ideal and negative INL means transition earlier than ideal [7]. It is measured in LSBs.

\[
INL = \frac{1}{2} V_{LSB}
\]

Fig. 6. Integral non-linearity [7]

II. Differential non-linearity (DNL)

It is measure of the maximum deviation from the ideal step size of 1 LSB. Results in narrow or wider code widths than ideal and can result in missing codes [6], [25]. DNL error is measured in LSBs [7].

\[
DNL = \frac{1}{2} V_{LSB} \Rightarrow \Delta V = \frac{1}{2} V_{LSB}
\]

\[
DNL = \frac{1}{2} V_{LSB} \Rightarrow \Delta V = 1.5 V_{LSB}
\]

Fig. 7. Differential non-linearity [7]

D. Aperture error

It is generated by a clock jitter and it is generated when digitizing a time variant signal. Imaging a sine wave signal, $X(t) = A \sin(2 \pi f 0 t)$ provided that the actual sampling time uncertainty due to the clock jitter is $dt$, error can be estimated as [5],

\[
dt < \frac{1}{2^q * \pi * f 0} \quad \text{………………… (3)}
\]

Where, $q$ = no of ADC bits.

The error is zero for DC, small at low frequencies, but significant when high frequencies. Clock jitter is caused by phase noise.

V. TYPES OF ADCS

There is a different way to implement an electronic ADC device is shown below:
A. Direct conversion or Flash ADC

It has a bank of comparators sampling the input signal in parallel. The comparator bank feeds a logic circuit that generates a code for each voltage range as shown in figure 8. It takes input parallel and as per the input voltage, comparator generates logic code and these logic codes are converted to digital data by an encoder [8], [10]. Direct conversion is very fast, capable of gigahertz sampling rates, but only 8 bits of resolution, since the no of comparators needed $2^n - 1$, doubles with each additional bit, requiring a large & expensive circuit [23]. It has large die size, a high input capacitance and high power dissipation. They are used for video, ultra wideband communication or other fast signals in optical storage, disk drivers, cable modems, CCD image systems, digital video and fast Ethernet systems [9]. They are used when a high sampling rate is required doesn’t matter about high resolution.

![Fig. 8. Flash ADC Architecture [8]](image)

Flash ADC is limited by only a few number of bits of resolution up to 8-bits because of the complexity of the circuits. [9] In a flash ADC the comparator only needs to be low offset and to resolve its input to a digital level; there is no linear settling time involved. Some flash converters require preamplifiers to drive the comparators. [10] Mohamed Shaker suggests a low power 1-GS/s 6-bit CMOS flash ADC. This design requires only few no of comparator and multiplexer to generate the required binary code. Using this multiplexer no of comparator is required less. This architecture can be extended to high resolution applications because of the simplicity of the circuit.

There is also solution for lower bits of resolution, with the use of two-step flash ADC. In two-step flash ADC conversion is done by two steps one is coarse flash ADC (MSB) and another is fine ADC (LSB). [25] In this case increase the no of bits of resolution and decrease the no of comparator is required from $2^n - 1$ to $2(2^n/2 - 1)$, so that power consumption is also decrease and increase the ADC performance. Hence this process is very easy and in future it is used for high resolution application. [11] Aamir Zijjo suggests a low power, low voltage for two step ADC. It uses a time interleaved sample and hold circuit along with 5-bit coarse ADC and 8-bit fine ADC. The 8-bit of fine ADC are generated with sufficient accuracy without using compensation by using a folding and interpolating ADC.

B. Successive Approximation ADC (SAR)

It uses a comparator to successively narrow range that contains the input voltage. [7-8], [25] At each successive step, the converter compares the input voltage to the output of an internal digital to analog converter (DAC). At each step in this process, the approximation is stored in a successive approximation register (SAR). As shown in figure 9 input analog voltages Vin is given to comparator, during the first clock pulse low-to-high transition, the MSB Q7 of the SAR is set. Then DAC generate analog equivalent voltage to Q7 bit, which is compared to Vin. If the comparator output is low, it means the DAC output Va>Vin, then the SAR clear its Q7 bit and set Q6 bit. If the comparator output is high, the DAC output Va<Vin then the SAR will keep the MSB Q7 set. This process is continue until both are equal Va = Vin. At that time SAR forces conversion complete & enable latch, so it takes data parallel as shown in figure 9.

![Fig. 9. SAR ADC Architecture [8]](image)

In a SAR ADC comparator determines one bit at a time from the MSB down to the LSB. This serial nature of the SAR limits its speed to no more than a few mega-samples per second (MSPS). SAR converters are available in resolution up to 16-bits. The slower speed also allows the SAR ADC to be much lower in power consumption. [12] This paper presents 1-V 8-bit 50 KS/s successive approximation analog to digital converter implemented in 1.2 μm CMOS. It is based on inverting op-amp configuration with biasing current added to the op-amp negative terminal.

C. Wilkinson ADC

It is based on the comparison of an input voltage with that produced by a charging capacitor. The capacitor is allowed to charge until its voltage is equal to the amplitude of the input pulse. Then the capacitor is allowed to discharge linearly, which produces a ramp voltage. At the point when the capacitor begins to discharge, a gate pulse initiated. The gate pulse remains on until the capacitor is completely discharged.
Thus the duration of the gate pulse is directly proportional to the amplitude of the input pulse [5], [24]. These ADCs have the best differential non-linearity (DNL).

D. Pipeline ADC

It uses two or more steps of sub ranging. First coarse conversion is done. In a second step, the difference to the input signal is determined with a digital to analog converter (DAC) [5], [14]. This difference is then converted finer, and the results are combined in a last step, as shown in figure 10. By combining the merit of the SAR & FLASH ADCs this type is fast, has a high resolution and only requires a small die size. Generally for high resolution and high speed this ADC is used. [25] A pipeline ADC employs a parallel structure in which each stage works on to a few bits of successive samples concurrently. This design improves speed at the expense of power and latency, but each pipelined stage is much slower than flash section.

Fig. 10. Pipeline ADC Architecture [14]

[13] The pipeline ADC requires accurate amplification in the DACs and interstage amplifiers, and these stages have to settle to the desired linearity level. Pipeline architecture provided resolution from 8 to 14 bits. [14] This paper presents capacitor error-averaging technique, updated with look-ahead decision and digital calibration for 14-bit pipeline architecture. This work achieves by using error-averaging amplifiers to eliminate errors resulting from capacitor ratio mismatch. [15] This paper presents a high speed single-channel pipeline ADC sampling at 2.4 GS/s. the high sampling rate is achieved through the use of fast open-loop current-mode amplifiers and the early comparison scheme.

There is also a time interleave architecture is used in which N different track-and-hold circuit is used along with N sub-ADCs are used. [16] Time interleaved architectures suffer from three impairments: offset, gain, phase skew mismatch between the banks of sub-ADCs. In this case N different clock phase generators are needed. Hence these are very complex process, so that different technique is used. [17] Sandeep and Michael suggest time interleaved architecture that eliminates the need to correct timing offsets and is scalable to high sampling rate. To eliminate timing skews, a Nyquist rate sampling switch is used, which is followed by subsampled, double-sampled time-interleaved sample and hold stages.

VI. CONCLUSION

Analog to digital conversion are reviewed in this paper using various techniques such as: flash or direct conversion, successive approximation, Wilkinson, pipeline, two-step, time-interleaved architecture. These different methods are used as per the requirement for the high speed, high resolution, less area, low power dissipation, low voltage etc. Some reference papers are listed below for more information.

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Efficient Wireless Data Transfer For Real Time Greenhouse Management Using Ieee 802.15.4/Zigbee Protocol

K.Kothavari, N.Praveen Kumar, S.SibhiPrasanna, S.Ramachandar, E. Vijay Antony, P. Pradeep

Abstract—This paper provides the practical solution for real time greenhouse management system using wireless data flow control, using labVIEW one can remotely monitor for each node assigned and the data can be acquired so with respect to the user provided set values, corresponding control instructions is transmitted form remote area/unit to control area. After the control action the node unit keeps on monitoring the chamber the status is updated and stored in spread sheet with the help of labVIEW. By this approach user possibly use proper resources which gives maximum yield throughout the year and a case study can be made by analysing the plant growth at different environment levels, these levels can be assigned and maintained by user.


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I INTRODUCTION

In our day to day life agriculture plays a vital role, the quality and cost are the major problem in food products, there is a greater demand for food product due to drastic population growth, our world population is expected to reach 9.1 billion people by 2050, Agricultural production demand increases rapidly (survey by MICCA). The environmental issues such as temperature, humidity, soil and pest are the factors affect the crop growth and productivity.

With the help of virtual tool such as LabVIEW, embedded with sensors, the proposed design is to measure, acquire and control user required parameter. An autonomous system contribute improved and constant yield.

II EXISTING SYSTEM ANALYSIS

To develop an efficient control system the concepts and ideas where obtained by analysing the papers, which gives idea on monitoring system[1], solar power energy harvesting[1] which helps to have an effective design, System gives idea on water management [2] which is compared in table 1.1. By referring the mentioned papers efficient control system is designed for real-time greenhouse.

A.Problem Identified

- Measured parameter are not controlled (Real time readings are observed)[1],[5].
- The storage depends on web server (Real time data directly transmitted for a central data base)[1],[7],[6].
- The data transfer rate as the distance increases[2].
- Loss of packets (For every 20 nodes, 8 nodes have complete loss of packet approx. 40% packet loss)[3].
- Time scheduling problem occurs (due to broadcasting, collision occurs)[4].
- Absence of precise prediction algorithm for calculating the risk index[3],[6].
- Time delay due to inefficient data transfer needs compression algorithm[3].

B.TABLE 1.1 SYSTEM COMPARISON
III PROPOSED SYSTEM

The greenhouse monitoring and control system based on WSN includes monitors the chamber and control actuators. Sensor nodes are deployed in greenhouse wherever, and preside periodic collection greenhouse environmental message and transmit to control centre, it is constituted by Indoor wireless unit, Outdoor wireless unit, Remote unit. These data are handled and analysed when control centre gains, then relevant decisions are made and send control message to greenhouse control unit by LabVIEW, which regulate greenhouse environment parameters to obtain best growth environment for crops.

- The control algorithm is developed for maintaining temperature, humidity, and soil moisture.
- The complexity is to Developing a greenhouse remote monitoring system that does not require manual changes with each crop change.

The proposed system approaches
- Monitoring
- Data acquisition
- Control

A. Node Unit

Each Electronic zone node(EZN) consist of PIC16F877A with the ZigBEE CC2205 transceiver use to transfer the temp, humidity and, soil wetness data with the help of LM35, HSY220 and conduction wires for soil wetness as shown in figure 1.4. LM35 have no normalization and it gives linear output, based on the conductivity the varies voltage level determined for different level of soil moisture and moreover comparing to wilting point the conduction rods placed for each one feet distance from ground which helps to identify the wetness with respect to depth. The collectively binded data are transmitted to PC unit.

![Figure 1.4 Block Diagram of Node Unit](image)

B. MONITORING

The initial step is monitoring every user supposed to monitor the status of the chamber unit, the number of chamber have each node and actuators to control. As shown in block (figure1.1) the sensors receives the analog input which corresponding voltage is generated as given below. Similarly each unit generates the voltage which is normalized value for output. These data where grouped

<table>
<thead>
<tr>
<th>Data</th>
<th>REFERENCE PAPER</th>
<th>Proposed system</th>
</tr>
</thead>
<tbody>
<tr>
<td>Technolog y</td>
<td>Ad-hoc</td>
<td>Wireless communicat ion</td>
</tr>
<tr>
<td>Hardware</td>
<td>wi-max</td>
<td>Mc system stc12le5410 ad (8051)gsm</td>
</tr>
<tr>
<td>Software</td>
<td>Simulating mac protocol</td>
<td>Visual basic</td>
</tr>
<tr>
<td>Features</td>
<td>Solar radiation wind speed and leaf wetness Centralized database.</td>
<td>Extinction of field area is possible</td>
</tr>
</tbody>
</table>

Table:

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<td>Technolog y</td>
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<td>Hardware</td>
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<tr>
<td>Software</td>
<td></td>
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<tr>
<td>Features</td>
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</tbody>
</table>
and transmitted with ZigBEEcc2205 (20m range). As shown in block diagram the sensors where interfaced in pin 2, 3, and 4 the corresponding pin configuration is shown in figure 1.5.

![Sensor Interfacing](image1.png)

**Figure 1.5 Sensor Interfacing**

The sensor output values where acquired and the corresponding test result of temperature sensor is shown in figure 1.6.

![Analog Output of Temperature Sensor](image2.png)

**Figure 1.6 Analog Output of Temperature Sensor**

The sensor output values where acquired and the corresponding test result of humidity and soil moisture sensor is shown in figure 1.7(a) and (b).

![Analog Output of Humidity Sensor](image3.png)

**Figure 1.7(a) Analog Output of humidity Sensor**

![Analog Output of Soil Moisture Sensor](image4.png)

**Figure 1.7(b) Analog Output of soil moisture sensor.**

These data where bundled and transmitted by 8 bit to the PC unit with the priority assigned to the controller and the values where obtained in LabVIEW as given below.

C.PC Unit

The obtained values from EZN where acquired to PC via USART serial communication, as shown in the figure 1.8. The each zone values are obtained these values are compared with the measured value.

![PC Unit for Serial Communication](image5.png)

**Figure 1.8 PC Unit for Serial Communication**

The assigned two zones where indicated in figure 1.9 and the initial values are assigned as shown in figure 1.10.
Further the vales obtained as shown in figure 1.11 is compared with the user provided data, here its processed with the sample data which is given detail in control unit.

D.Control Unit
The control unit provided with corresponding control algorithm, to trigger the actuators and to maintain the chamber environment as block shown figure 1.12.

Table 1.2 Values Obtained from Each Zone

<table>
<thead>
<tr>
<th>Temperature(°C)</th>
<th>Humidity(%RH)</th>
<th>Soil Moisture (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>24</td>
<td>64</td>
<td>10</td>
</tr>
</tbody>
</table>

Table 1.3 User Defined Value/ Required Value

<table>
<thead>
<tr>
<th>Temperature range (°C)</th>
<th>Humidity(%RH)</th>
<th>Soil Moisture (%)</th>
</tr>
</thead>
<tbody>
<tr>
<td>25-60</td>
<td>60</td>
<td>15</td>
</tr>
</tbody>
</table>

Table 1.4 Corresponding Control Action

<table>
<thead>
<tr>
<th>Blower Condition</th>
<th>Cooling Fan Condition</th>
<th>Sprinkler Condition</th>
</tr>
</thead>
<tbody>
<tr>
<td>ON</td>
<td>OFF</td>
<td>ON</td>
</tr>
</tbody>
</table>
E. Data Output
The two zone acquisition is done in this system which is implemented. In this, temperature values from the each zone are stored in the data base sheet in LabVIEW. The measured values are plotted and represented in the graphical format as shown in the figure.

F. Configuring with LabVIEW

For data logging the LabVIEW is configured as shown in figure 1.14 in such a way the each parameters are stored in excel sheet and graph is plotted as shown in figure 1.15.

This the final or net output for one sample of data, where the number of case study can be made for different crop unit in a chamber as shown in figure 1.17.
Where the proposed system is compared with the existing system as shown in table 1.5.

The system can be enhanced by adding the input parameters such as acquiring pH value from soil, the integrated pest management using image processing and the present remote monitoring can be enhanced by sharing data with network cloud using LAN protocol, it can be adopted to solar dryer chamber and also for outdoor field monitoring.

IV CONCLUSION

Table 1.5: Result Comparision

<table>
<thead>
<tr>
<th>S.no</th>
<th>Existing system</th>
<th>Developed system</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>System features half duplex communication</td>
<td>System features full duplex communication</td>
</tr>
<tr>
<td>2</td>
<td>Solar provides 10W, 0.67 Ah/day</td>
<td>Solar provided with 16W.</td>
</tr>
<tr>
<td>3</td>
<td>Data bit rate decreases with increase in bus rate(CAN)</td>
<td>Gives reliable data delivery in 20m(ZigBEE)</td>
</tr>
<tr>
<td>4</td>
<td>Does data base management</td>
<td>System have database management as shown in figure 1.16</td>
</tr>
</tbody>
</table>


REFERENCES


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BIOGRAPHY

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