**ELECRONICS & COMMINICATION ENGINEERING**

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Author : **PADAVALA AKHENDRA KUMAR**

Title of the thesis : **DESIGN, ANALYSIS AND OPTIMIZATION OF ON-CHIP**

**FRACTAL PASSIVE COMPONENTS FOR WIRELESS APPLICATIONS**

Guide : **Dr. N. BHEEMA RAO**

Degree : **Ph. D.**

Student ID. No : **714023**

**ABSTRACT**

Recent advances in the field of wireless communication demands high performance, low cost miniaturized wireless devices with low power consumption. Integrated passive components such as Inductors, Capacitors and Transformers have the potential to improve the performance of RF building blocks. The integration of passive components possess several challenges like design complexity, high-quality factor, small on-chip area and manufacturing costs. In this thesis, some of the challenges are addressed.

Passive components are generally designed using an EM simulator. The performance metrics of the passive components need a trade off between the cost of a specified technology and application. Fractal geometry has proven to be effective for the performance improvement of the passive components. In this thesis fractal geometry is adopted to reduce the foot print of the passive components.

Initially, in this thesis, fractal spiral capacitor is developed in single layer. The use of fractal geometry improves the lateral flux capacitance with the improved Q factor and self resonant frequency. The capacitance of the capacitor is further improved by taking the advantage of multilayer configurations. The proposed capacitor consists of two metal layers, each metal layer composed of two conductive metal tracks which are cross coupled. This structure effectively utilizes both the lateral and vertical flux capacitance. The sensitivity analysis of multilayer fractal spiral capacitor gives a trade off between electrical response and layout parameters.

Fractal inductors are developed by adopting two different fractal configurations. One is in the form of loop construction based on Hilbert and Omega based space filling curve and its mathematical analysis is also developed for inductance extraction. The concept extends to the spiral form to achieve high inductance per unit area. The Q factor and self resonant frequency of the structure is improved by taking the advantage of multilayer construction. The other is based on the modified Hilbert space filling curve. The proposed structure is having both structural and electrical symmetry to achieve high Q factor and self resonant frequency than the conventional Hilbert space filling curve. It is extended to multilayer layer to achieve high inductance per unit area. A novel shredded construction is applied to the lower metal layer to improve the Q factor of the stacked construction. The proposed inductor structures have better performance in terms of area reduction, Q factor and self resonant frequency for a specified inductance value. By using complementary split ring array based Energy band gap surface, the Q factor of the fractal inductor is enhanced. The left handed behavior and diamagnetism of the surface results in the improvement of Q factor values. EBG structures are designed for both on-chip and PCB based implementations.

An effective fractal inductor synthesis is proposed using improved feed-forward neural network. The developed ANN synthesis procedure generates layout parameters within tolerance limits. The synthesis results facilitate the designer to reduce the usage of EM simulator.

Two key RF building block applications, low pass filter and low noise amplifier are implemented to validate the proposed fractal passive components. The low pass filter is implemented based on fractal spiral capacitor and fractal spiral inductor in multilayer configurations. The fractal spiral capacitor occupies almost less than 50% of on chip area and the multilayer fractal spiral inductor achieves high Q factor resulting in low insertion loss miniature low pass filter. Similarly, the proposed multilayer inductor is used in the design of low noise amplifier that achieves a low noise figure and good input matching.

Some of the proposed passive components are fabricated on Printed Circuit Board for experimental validation. The structures are scaled up from m to mm and validated with the help of vector network analyzer. For all the models, experimental results are in good agreement with simulation results.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **ASHOK AGARWAL**

Title of the thesis : **AREA EFFICIENT RECONFIGURABLE ARCHITECTURES FOR**

**SAMPLE RATE CONVERSION IN SDR RECEIVERS**

Guide : **Dr. B. Lakshmi**

Degree : **Ph. D.**

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**ABSTRACT**

Due to recent advancements in wireless technology, there is a tremendous growth in the wireless industry and huge demand for high data rate applications. Over the past two decades, the focus of researchers has been to investigate new radio communication standards, such that the data rates as high as possible can be supported. This led to the changes in the radio communication standards at a faster rate. Radio communication systems when implemented as a hardware radio, the hardware becomes obsolete or needs to be redesigned with the changing radio standards. Solution to address this problem is Software Defined Radio (SDR).

A radio transceiver consists of Baseband (BB) processing stage, Intermediate Frequency (IF) processing stage and Radio Frequency (RF) processing stage. In an ideal SDR, all the three stages are implemented in software by placing data converters immediately after the antenna. Due to practical limitations of data converters, a practical SDR architecture is obtained by placing data converters at IF stage. In our work, we have focused on the design of IF stage of SDR receiver.

At the IF stage of SDR receiver, if the wideband signal is digitized at Nyquist rate the narrow band radio channels gets oversampled due to the phenomenon of pass band sampling. The signal processing carried out at this sampling rate leads to high power dissipation in the further stages of SDR receiver. Hence, a sample rate converter (SRC) is required to decimate the sample rate as per the specifications of radio standard into consideration. In SDR receivers, it is required to achieve SRC by programmable integer rate or programmable fractional rate or both. Integer rate SRC achieves SRC by large rates, while, the fractional rate SRC is required to make the signal sample rate power of two multiple of symbol rate of the radio standard. Hence, it is required to design a sample rate converter with reconfigurable SRC factors and tunable spectral characteristics while achieving minimum reconfiguration overhead and low hardware complexity.

Coefficient-less cascaded-integrator-comb (CIC) filters, a special class of filters proposed in the literature are suitable for achieving SRC by large integer factors as required in SDR receivers. These filters require less power consumption and can be reconfigured flexibly due to their coefficient-less architectures. CIC filters introduce gain droop in the pass band of interest and cannot achieve SRC by fractional rates. Hence, it is required to cascade a gain droop compensation and fractional rate interpolation filter.

In our work, we have considered four radio standards, WiMAX, WCDMA, CDMA2000 and GSM900 for multi-standard radio implementation with an IF frequency of 80MHz. We have employed CIC filters followed by compensation and interpolation filters designed based on discrete compensation and interpolation method, joint compensation and interpolation method and simulated for their frequency response in MATLAB. Compensation filters designed for one radio standard employing these methods require offline computations to support new radio communication standard.

In the present work, we make an attempt to eliminate the need of offline computations. We propose to apply singular value decomposition based variable digital filter (SVD-VDF) technique to design a joint compensation and interpolation filter. The SVDVDF consists of fixed coefficient sub-filters and multi-dimensional polynomials in terms of spectral parameters to reconfigure the spectral characteristics. Further ,the hardware complexity of proposed SVD-VDF based joint compensation and interpolation filter is reduced by proposing multiplier-less distributed arithmetic (DA) architecture for implementation of sub-filters in the SVD-VDF filter.

The functionality of the proposed reconfigurable SRC filter is verified by simulating the VHDL netlist and implementing on Kintex-7 XC7k325t-2ffg900 FPGA using Xilinx Vivado 2014.2 software. The performance of the proposed DA based architectures is computed by synthesizing using Synopsis Design Compiler with TSMC CMOS 90nm technology library. The performance comparison of these proposed DA-based architectures with the DA-based architectures available in the literature shows improvement in hardware complexity and power consumption.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **MUDASIR BASHIR**

Title of the thesis : **CMOS BASED ON-CHIP SMART TEMPERATURE SENSOR CIRCUITS FOR MILITARY APPLICATIONS**

Ph.D Guide : **Dr. P. SREEHARI RAO**

Degree : **Ph. D.**

Student ID. No : **714025**

**ABSTRACT**

Temperature monitoring is becoming progressively vital in electronic system designs. The scaling of transistor feature size in complementary metal-oxide-semiconductor (CMOS) technologies and increased integration density in system on chips (SoCs) has elevated the on-chip power density and junction temperature. The increase in chip temperature diminishes the lifetime, reliability and performances of SoCs, while increasing the issues like leakage current/power, interconnect resistances, delay and the overall cost. Therefore, temperature sensors are employed to provide key inputs to the temperature management system for keeping a check on SoC temperature variations. In applications where on-chip temperature is a concern, smart temperature sensor are placed at critical points or hotspots throughout the SoC to monitor and control the temperature. These smart temperature sensors should be energy efficient, accurate, having good resolution and compatible with CMOS technologies besides consuming less power and die area for the desired temperature range over the typical supply voltage and standard process variations.

The smart temperature sensors introduced in this thesis are designed for devices/equipments used for military applications, operating over a range of 􀀀55 °C to 125 °C. The temperature sensors exploit the thermal dependency of the threshold voltage of MOS transistors for temperature measurement. The sensors are fully CMOS based with dedicated signal conditioning circuitry and therefore can be also operated as stand-alone devices.

This thesis contributes three topologies of all-CMOS based on-chip smart temperature sensors for military applications, viz: 1) Current to frequency based temperature sensor (CFTS), 2) Voltage to frequency based temperature sensor (VFTS), and 3) Current mode dual slope based temperature sensor (CMDSTS). In CFTS, a frequency is generated from the current proportional to absolute temperature (PTAT) and then given to an asynchronous counter for digital conversion. It uses a two point on-chip calibration technique by employing a binary search algorithm for better accuracy. The VFTS operated in sub-threshold region of MOS transistors employs a frequency locked loop to generate a frequency from a voltage PTAT, which is given to an asynchronous counter for digital output. The VFTS requires one-point calibration, which is done by using binary weighted current sources controlled by dynamic element matching technique. Finally, a CMDSTS is introduced where a voltage is generated from PTAT current generator. A novel second order sigma-delta modulator is employed as a part of signal conditioning circuit for digital conversion. The CMDSTS also requires one-point calibration which is realized by the same technique as in VFTS. The low power operation of these sensors makes them suitable for SoCs and radio-frequency identification (RFID) tags used in military applications.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **GOVIND RAO DODDAMANI**

Title of the thesis : **Design and Realization of Novel Adaptive Digital Beam Former Architecture for Active Phased Array Radar**

Ph.D Guide : **Dr. T. KISHORE KUMAR**

Degree : **Ph. D.**

Student ID. No : **701144**

**ABSTRACT**

The concept of beam forming in phased array was implemented with analog or mechanical solutions. The present day need is to have every element level beam formation instead of a group of elements combined in the RF domain. Element level beam formation was a major challenge to realize in hardware. Due to this limitation researchers have developed sub array level beam formers for phased array application but this brings many limitations like roll off of the sub array pattern which causes a gain loss in the re-steered direction (Reduces the range of the radar) and produces grating lobes.

The digital systems have become powerful to carry out the huge number of tasks required for real-time digital beam formation. Today many applications are using beam forming to enhance the effective channel utilization both in frequency and space, more digital dedicated architectures are proposed for the parallel and pipelined processing for communication applications. With the help of reconfigurable system, the same hardware platform can be reutilised for different applications with different processing needs. This thesis designed a novel method of hardware design and realization of adaptive digital beam former for phased array Radar application.

This research work was carried out considering a typical case of sixteen element planar phased array. Initially multiple digital beam formation is achieved using the computation of weights off line and stored in the memory of the digital board and 4 and 8 receive beams were formed. The digital beam formation with fixed weights architecture is designed and functional simulation has been carried out. The same architecture is implemented on the Virtex-VI FPGA based hardware and multiple receive beams were formed.

Further the architecture development is extended to computation of adaptive weights online and a parallel pipelined architecture is designed. A survey of various algorithms has been carried out for adaptive weight calculation and QR-Decomposition based Recursive Least Square (QRD-RLS) is identified as most suitable. Since this algorithm is computationally complex and takes more time for the optimal weight calculation a modified algorithm is developed for online weight calculation. The Inverse QRD-RLS algorithm is a most efficient adaptive algorithm and from the hardware realisation point of view it is optimised. A systolic structure method can be employed to calculate the weights in a given time and is numerically stable compared to other traditional algorithms.

The most suitable algorithm for pipelined and parallel implementation architecture is the inverse QRD-RLS algorithm. For phased array radar applications optimal weights must be computed in a short time of the order of few micro seconds with very good accuracy. In such cases Inverse QRD-RLS is most suitable.

The novel architecture is developed for sixteen element planar phased array to form multiple receive beams for radar applications. Present day FPGA‘s are capable of concurrent processing, a three FPGA architecture is developed to form the adaptive beams simultaneously.

The outcome of this research work is realisation of generic, modular, scalable adaptive digital beam former architecture for a sixteen element planar antenna array configuration. This also can be extended for larger dimension in the phased array application.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **RAHUL SINGHA**

Title of the thesis : **SOME INVESTIGATIONS ON ULTRA-WIDEBAND**

**ANTENNA USING METAMATERIAL AND SUPERSTRATE**

Ph. D. Guide : **Dr. D. VAKULA**

Degree : **Ph. D.**

ID.No : **701351**

**ABSTRACT**

Wireless communication systems need broad bandwidth for faster connections among various wireless devices, where as narrowband and multiband systems have small bandwidth which limit their applications. The applications of wireless communications can be increased by adopting ultra-wideband technology. Ultra-wideband systems are designed to operate from 3.1 to 10.6 GHz for transmitting low power. Modern wireless communication devices need a compact broadband and low cost antenna. Wireless electronic system such as printer, mouse, smart watch, cell phone etc. require broad bandwidth with high data rate for high speed internet connection.

A compact UWB antennas with directional and omnidirectional characteristics are required for commercial, aerospace and defence industry applications and also for future complex wireless devices. Tapered slot antenna can be operated at broad bandwidth and hence are suitable for UWB directional applications. Vivaldi antennas which are end fire radiators are designed over UWB frequency band with various taper profiles like linear, exponential, elliptical and also corrugated along the antenna edges for performance improvement. In the thesis, the experimental results of the antennas are compared with the published literature. On the other hand, UWB antenna with omnidirectional characteristics are designed using printed monopole antenna with high gain.

Antenna designs are modified to improve the performance in terms of return loss, radiation pattern, gain, efficiency and group delay using superstrate and metamaterial. It has been established that by using fractal structure on the radiating patch, size of the antenna can be reduced with enhanced bandwidth and improved radiation. Moreover, it is also shown that dielectric superstrate can be placed on the fractal monopole antenna to enhance the gain of the antenna. The parallel-line structure metamaterial (MTM) is also used to enhance the performance of the tapered slot antenna and monopole antenna. Gradient refractive index metamaterial lens is designed by using different refractive index unit cell MTM. This helps to focus and make the radiated beam more directional. Hence, the gain of the antenna can be increased. The proposed metamaterial has low material loss covering UWB frequency band and is easy to fabricate. The band notch characteristics are also included to avoid a serious electromagnetic interference within UWB spectrum. The band notch characteristics at WLAN and ITU band are implemented by using split ring resonator and electric field coupled resonator in UWB tapered slot antenna. Due to the these MTM, the gain of the antenna is also slightly decreased at other frequencies for harmonic effects. The S-shaped unit cell meta material are also used to enhance the gain of the antenna. Antennas are simulated using three dimensional High Frequency Electromagnetic Field Simulation (HFSS) Simulator and they are fabricated, tested and also compared with simulation results.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **P. RAKESH**

Title of the thesis : **DEVELOPMENT OF NOVEL SPARSE-AWARE**

**ALGORITHMS FOR ADAPTIVE SYSTEM IDENTIFICATION**

Guide : **Dr.** **T. KISHORE KUMAR**

Degree : **Ph. D.**

Student ID. No. : **701350**

**ABSTRACT**

Adaptive filtering algorithms play a key role in adaptive signal processing, especially for applications where real-time estimation of some unknown parameters is required.

In this thesis, the significance of adaptive filters in system identification configuration is considered. Adaptive echo cancellation and channel estimation are the two prominent communications applications in system identification configuration.

In order to determine the transfer function estimate for an unknown digital or analog system, one can use the adaptive system identification (ASI). System identification describes the task of identifying an existing unknown system and adaptive filters are widely used for this application. In many scenarios of system identification, the impulse response of underlying system is presumed to be sparse which means most of its coefficients are zeros (inactive) and have few non-zero values (active). The basic methodology behind the sparse system identification is to make use of the prior sparse information to improve its filtering/estimation performance.

The traditional system identification algorithms are generally sparsity agnostic in nature viz. they are unaware of the underlying system sparsity which makes their application impractical for system identification. In order to exploit system sparsity, sparse adaptive filters are extensively used. Sparse-aware adaptive filtering algorithms offer improved performance. Hence, in this thesis, we consider the development of novel sparse adaptive algorithms for system identification. This thesis comprises of four parts:

1. Combinational approaches of adaptive filters for sparse system identification.
2. Sparse adaptive algorithms based on Lyapunov Stability for system identification.
3. Robust sparse system identification algorithms for impulsive noise environments.
4. Complex domain sparse adaptive algorithms for system identification.

In the first part, utilizing the benefits of combining two adaptive filters through a mixing parameter, we propose an affine combination of two Improved Proportionate Normalized LMS (IPNLMS) filters. Further, we also propose an affine combination of Reweighted Zero Attracting-NLMS (RZA-NLMS) and NLMS algorithm for system identification with variable sparsity. The combination approach provides the robust solution to alleviate the convergence speed vs steady-state error trade off, as well as to efficiently increase the filter robustness to time varying sparsity of the system.

In the second part, we consider the Lyapunov theory-based adaptive filter (LA) which offers to improve the convergence and stability, and also overcome the problems faced by gradient descent-based adaptive filtering techniques. In order to address system sparsity, the Zero-Attracting Lyapunov Adaptation algorithm (ZA-LA), the Reweighted Zero-Attracting Lyapunov Adaptation algorithm (RZA-LA) and an affine combination of the LA and ZA-LA adaptive filters are proposed. We show that the proposed sparse Lyapunov algorithms outperform the Least Mean Square (LMS) algorithm and its sparse counterpart (ZA-LMS and RZA-LMS) for both white input and colored input cases in terms of Mean Square Deviation (MSD) and Mean Square Error (MSE). The proposed affine combination filter is also robust in identifying the system with variable sparsity.

In the third part, we investigate the estimation performance of adaptive algorithms under the impulsive noise conditions. The novel sparse algorithms based on high-order error power (HOEP) criterion i.e., Normalized Least Mean Absolute Third (NLMAT) are proposed to mitigate the adverse effects of impulsive noise and to utilize the sparsity phenomenon effectively. Modified Least-Mean Mixed-Norm algorithm which is based on sigmoid function (SLMMN) is also developed to achieve robust performance against impulsive noise and the corresponding sparse SLMMN algorithms are proposed in the sparse system identification context.

In the fourth part, we discuss the complex domain sparse adaptive filter algorithms by incorporating different sparse penalty terms into the affine projection normalized correlation algorithm (APNCA). The proposed algorithms address the system sparsity as well as robustness against impulsive noise and achieves faster convergence for a correlated input.

All the proposed algorithms in this work are tested by extensive computer simulations. The results demonstrate significant performance improvement in terms of convergence rate, robustness against impulsive noise and steady-state error.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **M. RANJEETH**

Title of the thesis : **PERFORMANCE ANALYSIS OF COOPERATIVE SPECTRUM SENSING NETWORK OVER VARIOUS FADING CHANNELS**

Guide : **Dr. S. ANURADHA**

Degree : **Ph. D.**

Student ID. No. : **701424**

**ABSTRACT**

The demand for radio spectrum has dramatically increased over the past decade due to the proliferation of wireless services and applications. According to current static spectrum allocation policy, almost all frequency bands of radio spectrum has allocated and that there is a shortage of spectrum for new wireless services. On the other hand, actual measurements of radio spectrum usage have shown that most of the licensed spectrum is largely underutilized. Cognitive radio (CR) technology is considered as a promising technology to increase the efficiency of spectrum usage by allowing the secondary (unlicensed) users to opportunistically access the allocated spectrum of primary (licensed) users. To enable the secondary users (SUs) and to utilize the underutilized spectrum, they need to make the necessary observations about their surrounding radio environment. Therefore, spectrum sensing (SS) is required by the SUs to learn about the activities of primary users (PUs). However, the performance of spectrum sensing is limited due to multipath fading and shadowing effects present in the nature which are the fundamental characteristics of wireless channels. To overcome these challenges, cooperation among SUs is required to perform the spectrum sensing which is known as cooperative spectrum sensing (CSS) technique. The aim of this thesis is to increase the detection probability value and efficient utilization of radio spectrum using different techniques in cooperative spectrum sensing network when it is affected by various fading environments.

The influence of fading effect limits the detection probability of PU. So, soft data fusion rules called diversity techniques are used to improve the detection performance when CSS network is affected by various fading effects. Diversity techniques improve the received signal SNR and uses the multiple antennas to receive the signal strength. Initially, average detection probability values are calculated using different diversity techniques at fusion center (FC) when CSS network is affected by various fading environment to know which diversity technique gives higher detection probability. The detection probability values are calculated using various diversity techniques such as selection combining (SC), square law combining (SLC), square law selection (SLS), maximal ratio combining (MRC), and equal gain combining (EGC) over different fading environments like Rayleigh, Rician, Nakagami-m, Weibull, and Hoyt fading effects. The performance is evaluated using the conventional energy detector (CED) and single antenna at each CR in CSS network.

Next, due to the fading effect in the environment, some of the CR users are deeply affected, and sensing information associated with these CRs are not transferred to the FC accurately. So, it is better to eliminate these CR users; this can be done with the help of censoring schemes (Rank based and Threshold based censoring schemes). Two censoring schemes are used individually in reporting channel of CSS network when it is affected by Rician, Hoyt, and Weibull fading environments. The novel expressions for estimation error, its mean, and variance expressions are derived under various fading effects. The missed detection probability and total error values are calculated in various fading environments using the censoring schemes in CSS network.

Further, the performance is evaluated with proposed CSS network which is equipped with multiple antennas and an improved energy detector (IED) scheme at each CR over various fading channels. Selection combining scheme is also used at each CR to receive the binary decisions about the PU from an IED technique using multiple antennas and selects the better detection value of PU. The sensing information about the PU is passed to FC through the reporting channel (R-channel). Final decision is made at FC using different fusion rules such as OR-Rule, AND-Rule. The novel expressions of missed detection probability using multiple antennas and an IED scheme at each CR for different fading channels are derived. The optimized performance of proposed CSS network is achieved by optimizing its network parameters such as a number of CR users (N), the threshold value (λ), and arbitrary power of received signal (p). The novel optimized expressions of CSS network parameters such as number CR users (𝑁𝑜𝑝𝑡), threshold value (𝜆𝑜𝑝𝑡), and arbitrary power of received signal (𝑝𝑜𝑝𝑡) are also derived for various fading channels using single and multiple antennas at each CR.

Finally, average channel throughput (𝐶𝑎𝑣𝑔) and network utility network utility function (NUF) performances are evaluated using the proposed CSS network over Rayleigh and Weibull fading channels. The performance is evaluated using the different fusion rules such as k=1+n and k=N-n rules at FC. An optimal number CRs are calculated to maximize the average channel throughput value for different fusion rules. 𝐶𝑎𝑣𝑔 and NUF performances are improved by using multiple antennas and an IED scheme at each CR in the proposed CSS network.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **REBELLI SHASHANK**

Title of the thesis : **SIGNAL INTEGRITY ANALYSIS OF ON - CHIP INTERCONNECTS USING MRTD**

Guide : **Dr. N.BHEEMA RAO**

Degree : **Ph. D.**

Student ID. No. : **715045**

**ABSTRACT**

With the advent of technology scaling, the efficiency of the interconnects effect  
 the overall performance of the circuit. Although active devices mostly benefited from scaling, the performance of intermediate and global interconnects has degraded because long interconnects do not scale with the technology. Apart from power dissipation and overshoot issues, signal integrity issues such as propagation delay of long interconnects become a bottleneck in high-speed operation of ICs. Also, functional/dynamic crosstalk result in malfunctions in the circuit leading to reliability problems. Hence, there is a great demand for estimation of propagation delay and crosstalk noise of coupled interconnect lines in the early stages of VLSI design.

This thesis focuses on the development of a novel time-domain numerical method with significant numerical dispersion characteristics based on the wavelet scaling functions to address the signal integrity issues of on-chip interconnects. The multiresolution time domain (MRTD) model with its unique features is tailored for modeling VLSI interconnects. To build further credence to this and its profound existence in the recent state-of-the-art, simulations for inclusive crosstalk noise, on driver-interconnect-load (DIL) system, using the MRTD model and conventional finite-difference-time-domain (FDTD) model are performed.

Initially, in this thesis, an attempt is made to derive an MRTD scheme for two coupled copper (Cu) interconnect lines driven by the linear resistive driver in 130 nm CMOS standard process to compute the effect of coupling parasitics (i.e., coupling capacitance and mutual inductance) on peak crosstalk noise and propagation delay. For different  
 values of coupling parasitics, the variation in peak crosstalk noise and propagation delay is observed and a comparison is done between the obtained results with those of the conventional FDTD model with respect to HSPICE simulation results. Moreover, variation in accuracy of the proposed MRTD model for a range of frequencies is observed and encouragingly it is found that approximately 100 % accuracy is maintained for a broad frequency range although a slight perturbation does exist within a short range of frequencies.

However, in DIL systems, replacement of non-linear CMOS driver with a simple linear resistor leads to a discrepancy in the results as about half of the operating time of MOSFET is in the saturation region, whereas the other half is divided between the cuto  
 and the triode regions. Therefore, the proposed MRTD model is extended to include the non-linear characteristics of the CMOS driver in the DIL system for 32-nm technology node. The non-linear CMOS driver is analyzed by employing the n-th-power law model. For the robustness of the model, a different number of test cases in terms of input transition time are considered and the peak crosstalk noise and peak noise timing are also computed for the two coupled Cu interconnects. Further, the model is extended to three mutually-coupled Cu interconnect lines.

Further scaling of interconnect dimensions have made surface scattering and grain boundary scattering more prominent, resulting in increased resistivity of Cu material. Therefore, the requirements of novel material as VLSI interconnect has increased. In recent times, carbon nanomaterials such as carbon nanotubes (CNTs) and graphene nanoribbons (GNRs) act as the most promising candidates proposed as a substitute for Cu interconnects in advanced VLSI circuits. Thus, the proposed MRTD method is extended to analyse the inclusive crosstalk effects in CMOS gate driven two and three mutually-coupled MWCNT interconnects at 32-nm technology node. It is observed that a peak overshoot/undershoot occurs in the response of line 2 (victim line) as the conventional FDTD method has higher dispersion errors. Nevertheless, the numerical dispersion properties in MRTD model acts as an added advantage over the conventional FDTD model for achieving better accuracy. Finally, it is concluded that the proposed MRTD method is in good agreement with HSPICE simulations and dominates the conventional FDTD method. Furthermore, the validation of the proposed model with future selective validation (FSV) proves its accuracy and e efficiency for analyzing the crosstalk e effects in mutually coupled MWCNT interconnects.

This thesis shows that the proposed MRTD method is more time e efficient than HSPICE, although the elapsed CPU time of the proposed MRTD method is higher than the conventional FDTD method, due to an increased number of iterations for better accuracy. Hence, there exists a trade-o between simulation time and accuracy. The analysis has been carried out on two coupled and three-coupled interconnects, but can also be extended to N-mutually coupled interconnects.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **SHRAVAN KUMAR BANDARI**

Title of the thesis : **PERFORMANCE ANALYSIS OF GFDM FOR 5TH GENERATION**

**CELLULAR COMMUNICATION**

Guide : **Dr. V. Venkata Mani**

Degree : **Ph. D.**

Student ID. No. : **701356**

**ABSTRACT**

Over a last few decades, there is an enormous increasing interest in the field of wireless communication with continuous emerging demand for new services and applications in day-to-day daily life. Next generation wireless networks should have the capability to connect anything, anytime, anywhere and anypath with several key factors. This includes high data rates ( 10Gbps), high connection density ( one million (1M) connections per km2), massive security and safety and ultra low latency (1ms). Foreseeing the prerequisites of the future applications/services like Tactile Internet (TI), massive machine type communication (MTC), Internet of Things (IoT) and smart cities, high investments are made for developing the next generation (5G) physical layer multicarrier waveform. This dissertation will focus on several aspects of such a multicarrier scheme which serves the future human needs. Although there are many issues that can be addressed in the context of 5G, it is vital to choose a more reliable modulation scheme. One such multicarrier waveform is Generalized Frequency Division Multiplexing (GFDM) which serves the future wireless goals. Due to the impairments caused by multipath reactions, terrain conditions, scattering from different obstacles and diraction, the signal that has been transmitted may gets attenuated along with multiple copies of the transmitted signal being received at different time intervals. Hence it is important to study the channel fading e ects on the overall system performance. In this thesis we first express the signal processing steps of this pulse shaped exible GFDM waveform along with performance analysis under various fading channels. Accordingly, the performance of the developed system model is evaluated by deriving analytical expressions for probability of error over different fading environments like Nakagami-m, Rician- K, Nakagami-q (Hoyt), Weibull-v channels and Log-Normal shadowing is included. The effect of each fading parameter (m, K, q, v), filter roll-o factor (α) and modulation order () are examined. Due to inherent non-orthogonal nature of GFDM, there is a considerable degradation in the system performance. Consequently, investigations are carried to improve the orthogonality of the conventional GFDM system by proposing pulse shaping lter like discrete prolate spheroidal sequences (DPSSs) also known as multi-tapers. In doing so, an additional benefit of lower out of band radiation (OOB) is also achieved. Furthermore, for the proposed system, symbol error rate (SER) is computed over both additive white Gaussian noise (AWGN) and Rayleigh channels. In spite of signi ficant prevalence, the high peak-to-average power ratio (PAPR) which stems out from the superposition of huge number of subcarriers is inevitable in GFDM. Hence, investigations are carried out to reduce the amount of PAPR by incorporating several PAPR reduction schemes like Partial Transmit Sequence (PTS), Selected Mapping (SLM) and Zadoff  
-Chu sequences highlighting the spectral properties as well as emphasizing the complexity analysis. In addition, we propose a hybrid Walsh- Hadamard precoding with both SLM and PTS PAPR reduction techniques to further reduce the PAPR level at the transmitter side. The Walsh-Hadamard kernel which is a square wave with values f1g helps to reduce the PAPR without introducing distortion, compared to clipping based methods. Using these proposed schemes the results show a superior PAPR reduction improvement when compared to the conventional SLM and PTS techniques. Further improvement in Peak-to-Average-Power Ratio (PAPR) reduction can be done by replacing conventional Fourier transform used in traditional GFDM system by wavelet transforms, which is one of the key contribution of this work. To further improve the SER performance some modifications have been proposed. Of them primarily, exploiting the near orthogonality nature of o  
set quadrature amplitude multiplexing (OQAM), an improvement in both spectral efficiency and bit error rate performance is obtained upon comparison with the traditional GFDM system. Later, spatial diversity at the transmitter side is proposed under Rayleigh fading channel with the help of derived analytical expressions. At the same time different  
 channel estimation techniques have been discussed using pilot symbols as a reference signal and validated results with the support of the mathematical expressions derived for the channel estimation methods that has been investigated. Lastly, convolution codes are used to decrease the effect of environmental interferences for ensuring a reliable communication. Eventually, the aforementioned modifications enhances the performance of conventional GFDM system.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **SUDEEP SURENDRAN**

Title of the thesis : **DEVELOPMENT OF NOVEL PERCEPTUAL**

**SIGNAL SUBSPACE SPEECH ENHANCEMENT**

**ALGORITHMS FOR LOWDISTORTION AND COLORED NOISE REDUCTION**

Guide : **Dr. T. KISHORE KUMAR**

Degree : **Ph. D.**

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**ABSTRACT**

Speech enhancement is the improvement in the quality and the intelligibility of noise corrupted speech signals by using various signal processing techniques. It is commonly used as a pre-processing block in a lot of applications like automatic speech recognizer, various communication systems etc.

This thesis focuses on the development of novel speech enhancement algorithms based on signal subspace method to address the issues of signal distortion and residual noise in enhanced speech. An efficient noise estimation algorithm is employed in subspace approach to improve the performance of the enhancement algorithm. Algorithms are also developed to handle the case of colored noise effectively. The noise reduction algorithms are based on the eigenvalue decomposition, which is a robust and widely used computational tool for noise suppression.

Speech enhancement can be achieved using various techniques like spectral subtraction, adaptive Wiener filtering, model based methods etc. Conventional methods like Wiener filter, spectral subtraction etc. have been widely used for enhancing speech because of their simplicity and ease of implementation in single channel systems, but suffer from the production of musical noise after enhancement and is one of their major drawbacks. Signal distortion is considered as another important issue in speech enhancement. Also, most of the methods give a low performance in colored noise scenarios and hence fail in real life environments. Therefore, there is a need to develop speech enhancement algorithms with low signal distortion and residual noise, which is robust to all the noise scenarios that naturally exist.

Signal Subspace Approach of Speech Enhancement (SSASE) has given promising results in this direction, compared to the other existing techniques. SSASE initially separates the noisy speech subspace into noise subspace having noise elements alone and speech subspace having speech plus noise elements. To obtain the enhanced speech, the noise subspace is removed completely and the speech subspace is filtered to remove the remaining noise elements.

Using perceptual features, the masking properties of the human auditory system which exploits the fact that the audible threshold for one sound is raised by the presence of another sound, would make it possible to remove the audible noise elements alone and retain the inaudible noise components. This would avoid unnecessary filtering of the desired speech components, reducing signal distortion. In order to reduce residual noise, a signal estimator has to be designed which would reduce the residual noise without causing much signal distortion. Also, the performance of any speech enhancement algorithm greatly depends on the efficient estimation of the noise present in the noisy speech. The robustness of speech enhancement systems to any kind of noise could be achieved by employing such a noise estimation algorithm. To develop an SSASE algorithm to handle colored noise which is more common in real life environments, the use of techniques like oblique projection, which has the properties which could separate colored noise, would be the most appropriate.

The use of simultaneous and temporal masking together with variance and Signal to Spectral Distortion Ratio (SSDR) normalized post-filtering in SSASE showed a reduction in signal distortion compared to the conventional enhancement methods. To reduce the production of residual noise, a modified Spectral Domain Constrained (SDC) estimator is employed and the gain of the filter matrix is made to adaptively change according to the abrupt changes in the signal and the masking threshold. This reduces the generation of residual noise, making the output more intelligible. The use of a novel noise estimation algorithm based on Speech Presence Probability (SPP) in SSASE has improved the performance of the system over the use of the inefficient discontinuous noise estimation algorithms. To deal with the case of colored noise, the use of oblique projectors in SSASE to separate the colored noise elements from the noisy speech after the removal of additive noise using orthogonal projectors is proposed to obtain enhanced speech. Further, the oblique projection approach is extended to cepstral domain for better speech enhancement.

The results in terms of spectrograms, waveforms, objective measures and subjective measures show the superiority in the performances of the proposed algorithms compared to the existing speech enhancement techniques. The spectrograms and waveforms show that the proposed algorithms remove noise without causing much distortion. Objective and subjective measures also show high quality and intelligibility of the proposed methods.

This thesis shows the effectiveness and robustness of the developed signal subspace based speech enhancement algorithms. The superiority of the developed algorithms over the existing methods is evident from the waveforms, spectrograms, objective measures and subjective measures presented.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **SUMAN NELATURI**

Title of the thesis : **INVESTIGATIONS ON METAMATERIALS BASED DUAL**

**AND WIDE BAND MICROSTRIP PATCH ANTENNAS**

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Degree : **Ph. D.**

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**ABSTRACT**

Studies on metamaterials based microstrip patch antennas are carried out for dual band and wide band operation. Metamaterials like CSRR, MUC, ENGTL VIAs and RIS are used to design compact dual band antennas and HIS is used to design miniaturized wide band antennas with linear polarization at all the bands. The edges of the square patch are replaced with fractal curves such as Minkowski, Koch, Poly and Semi circled to achieve circular polarization at patch mode band.

Initially the proposed antennas are designed using HFSS Electromagnetic Simulator and later simulation results are verified experimentally. A new class of compact dual band patch antennas based on CSRR is designed with good impedance and CP bandwidth at patch mode band and less bandwidth at left hand band. Another class of dual band patch antennas are investigated based on MUC which are producing good impedance bandwidth at both the bands. The third class of dual band patch antennas studied are based on ENGTL VIAs which are responsible for good return loss bandwidth at both the bands and good CP bandwidth at patch mode band. The fourth class of dual band patch antennas demonstrated is based on RIS which are compact in size and are able to produce good return loss bandwidth at both the bands and the CP bandwidth at patch mode band.

The antennas mentioned above all operate at 2.4 GHz and 3.4 GHz frequencies and are useful for Wi-Fi and Wi-MAX applications. The fifth class of patch antennas is for wide band operation based on HIS working at 2.4 GHz for Wi-Fi application. The obtained impedance bandwidth is 15% and the 3-dB axial ratio bandwidth is 4%.

The dual band and wide band antennas proposed in this thesis provide good impedance bandwidth at both the bands and good circular polarization bandwidth at patch mode band. These antennas are best suitable for handheld gadgets which support wireless applications. The ideas given in this thesis are useful to the design engineers who are working on compact dual band and wide band microstrip patch antennas.

**ELECRONICS & COMMINICATION ENGINEERING**

Author : **SUSEELA VAPPANGI**

Title of the thesis : **PERFORMANCE ANALYSIS OF MULTICARRIER AND**

**MULTIPLE ACCESS SCHEMES COMPATIBLE WITH IM/DD**

**SYSTEMS FOR VLC**

Guide : **Dr. V. VENKATA MANI**

Degree : **Ph. D.**

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**ABSTRACT**

During the last decade, the rapid proliferation of mobile devices created an everincreasing need for high speed data access. This unintermittedly manifested the present radio frequency (RF) based wireless communication systems to suffer from spectrum congestion, limited data rates, huge increase in latency, decrease in reliability of services and interference. Concurrently, indoor illumination has been revolutionized by solid-state lighting, where light emitting diodes (LEDs) have been emanating as potential technology to replace current incandescent and fluorescent lamps due to their distinguished advantages such as long lighting hours, high efficiency, small size and low power consumption. Persuaded by this striking augmentation of LED technologies and in order to alleviate the RF spectrum scarcity problem, visible light communication (VLC) is arousing invariably tremendous momentum in both academia and industry. The noteworthy features offered by VLC like its worldwide usage, huge, license-free bandwidth and enhanced security facilitated VLC to evolve as a promising candidate to complement the conventional RF counterpart. The stupendous notability of VLC to facilitate illumination and communication contemporaneously by exploiting illuminating devices like LEDs will definitely enable VLC to emerge as an intriguing alternative to traditional RF-based wireless communication.

Foreseeing the future demands of several applications, this dissertation focuses on several aspects of VLC which are vital to be addressed to render high data rate communication. In spite of significant amount of prevalence, the low modulation bandwidth of white phosphorescent LEDs hinders the high data rate communication. Therefore, to boost the data rates, VLC adopts the robust multicarrier modulation technique like orthogonal frequency division multiplexing (OFDM) due to its inherent advantages like combating inter symbol interference, being resilient to frequency selective fading channel effects, simple single tap-equalization techniques, etc. The non-coherent emission characteristics of LEDs make intensity modulation (IM) and direct detection (DD) as the most viable modulation scheme for VLC. This is obvious because the data is transmitted by modulating the intensity of light. However, the traditional OFDM as used in RF cannot be directly employed in VLC due to the constraint that the transmitted signal has to be both real and positive-valued i.e., unipolar. It is apparent that inverse fast Fourier transform (IFFT) and fast Fourier transform (FFT) play a vital role for enabling modulation and demodulation in OFDM. When the incoming stream of data is mapped by employing complex constellation techniques like quadrature amplitude modulation (QAM), the output of the IFFT will be a complex and bipolar signal. Since, light intensity cannot be negative and complex, the input to the IFFT is constrained to satisfy Hermitian Symmetry criteria for yielding a real valued signal while dealing with optical OFDM. However, in doing so, only half of the subcarriers are utilized for the transmission of the data because the rest half are flipped complex conjugate versions of the previous ones. Therefore, this leads to a decrease in throughput and moreover incurs additional digital signal processing for the computation of Hermitian Symmetry.

Consequently, this entails to exploit real transformation techniques like discrete cosine transform (DCT), discrete sine transform (DST), and discrete Hartley transform (DHT), etc for obtaining a real-valued signal. In order to yield a positive-valued signal, DC-biased optical OFDM (DCO-OFDM), asymmetrically clipped optical OFDM (ACOOFDM) are considered in this work. In general DCO-OFDM consists of adding a DC bias to the time-domain signal in order to attain a positive valued signal. In specific, this addition of DC bias has made DCO-OFDM as a power inefficient scheme. ACOOFDM overcomes this drawback by avoiding the addition of DC bias but, this is at the expense of reduced throughput. Due to real and positive nature of the transmitted timedomain signal, firstly we derive the mathematical expressions for the time-domain signal formats in accordant with IM/DD systems for VLC. Taking into consideration the dispersive VLC channel environment in an indoor room environment, sensing and estimation of the channel is necessary. This work analyses the performance of comb-type pilot arrangement aided channel estimation for diff erent multicarrier transmission systems like DCO-OFDM, ACO-OFDM, DHT-based ACO-OFDM and Fast-Walsh Hadamard Coded Modulation (HCM), DCT-based DCO-OFDM over dispersive VLC channel. In addition, for the first time this work proposes a DST-based DCO-OFDM and ACO-OFDM system and evaluates the bit error rate (BER) performance of the developed system over both additive white Gaussian noise (AWGN) channel and diffused optical channel environment. Various channel estimation techniques like least square (LS), minimum mean square error (MMSE), different interpolation techniques like linear, spline and low-pass interpolation schemes are evaluated and compared for the aforementioned multicarrier systems. In addition, Cramer Rao Lower Bound (CRLB) is derived for the channel estimation error.

Furthermore, this thesis addresses a vital aspect like peak to average power ratio (PAPR) which arises due to the superimposition of huge number of subcarriers in optical OFDM. This aspect is fuelled with the limited dynamic range of LEDs, where it leads to clipping of the time-domain optical OFDM signal which doesn't t within the linear range of LEDs. Hence, investigations are carried to reduce the amount of PAPR by imposing several PAPR reduction techniques like Spreading technique, Partial Transmit Sequence (PTS), clipping and filtering. Exploiting the advantages of real trigonometric transform like DCT and DST, this work proposes DCT-spread and DST-spread technique to reduce the amount of PAPR in DCT/DST-based DCO-OFDM system. Upon comparison with other PAPR reduction schemes, the spreading technique significantly reduces the amount of PAPR. Additionally, in order to reduce the amount of PAPR in a multicarrier based system, this thesis expedites single carrier frequency division multiple access (SC-FDMA) and derives the mathematical expressions for different mapping strategies like optical localized frequency division multiple access (OLFDMA), optical interleaved FDMA (OIFDMA) and compares the PAPR performance with optical orthogonal frequency division multiple access (OOFDMA) for both DCT and DST based multiple access systems. The simulated results emphasize that among the multiple access schemes, PAPR reduction in OIFDMA is superior than OLFDMA and outperforms OOFDMA in both DCT/DST based systems.

Finally, this work addresses the performance of the developed multicarrier and multiple access systems in the presence of frequency offset (FO) and symbol time offset (STO) and furthermore, derives the mathematical analysis for different sorts of interferences like inter carrier interference (ICI), multi user interference (MUI) emanating in the uplink scenario in DFT-based and DCT-based DCO-OFDMA systems. This is a vital issue which needs to be addressed because, the presence of these offsets hinders the detection capability of the desired subscriber signal in the presence of multiple users. From the simulated results it can be evidenced that the BER performance of the proposed system deteriorates as the sensitivity of the offsets increases. Hence, in order to effectively estimate these offsets, this thesis imposes synchronization algorithms like Classen, Moose, Training symbol-assisted, Maximum correlation and Minimum difference algorithms. Additionally, CRLB is derived for the estimation of FO and STO and is verified through simulation results. From the simulation results, it can be evidenced that CRLB attains the minimum mean square error (MSE) upon comparison with other algorithms.

The major focus of this thesis is to derive the analytical expressions for different time-domain signals for both multicarrier and multiple access schemes which are fulfilling the requirements of cost-effective IM/DD systems for VLC. For the provision of reducing the computational complexity, this work analyses the performance of DCO-OFDM and ACO-OFDM system based on real signal transformation techniques like DCT, DST and DHT in terms of BER, MSE and PAPR and compares with the traditional HS imposed FFT-based optical OFDM. Even though several variants of OFDM have been proposed in the literature, the main motive behind employing DCO and ACO-OFDM is that majority of the variants are a combination of DCO and ACO-OFDM. Moreover, the other variants involve a complexity in the design of receiver. The proposed system models and the mathematical analysis carried in this work will enable VLC to be envisaged as a potential complementary to RF-based wireless communication.