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Microelectronics, VLSI & Display Technology

RF Microwaves and Photonics

Signal Processing, Communication & Networks

Power & Control

Title : ***Solution processed vertical organic transistor: Simulation, fabrication and characterization***

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Supervisor : ***Iyer S Sundar Kumar&Mohapatra Yashowanta Narayan***

Abstract

Recent developments in organic thin film transistors are proving to be useful in many applications. Though OTFT provides advantages of low cost and low temperature processing, the conventional lateral structures with large channel length suffer from high voltage range of operation, low mobility and low speed. To overcome some of these drawbacks of lateral OTFT, there is a search for convenient vertical structures using organic semiconductors. The focus of this work is mainly to establish a base-line process for solution processable vertical organic transistors so that challenges for mounting further developments can be identified. Specific aims of this work has been (a) to fabricate a reference OTFT using standard bottom gate lateral geometry with pentacene as the organic semiconductor; (b) to study mechanism of proposed vertical organic transistor through simulation so as to help experimental design; and (c) to set-up a base-line process for fabrication of VOT so that challenges in optimization can be isolated. The simulation of the device structure of VOT using ATLAS simulator demonstrates clearly that the operation of the device is not based on the depletion width in perforated gate of the device as is normally assumed. It is the potential distribution around metal grid which controls current streamlines which in turn effects the threshold voltage for space charge limited current. The ability of the gate to vary the threshold voltage for SCLC limited operation is central to the mechanism. However, simulations do not predict occurrence of saturation in output characteristics for designs considered. A baseline process for fabrication of a vertical organic transistor (VOT) was developed. The process includes spin-cast P3HT, and an Al grid pattern obtained using etching of polystyrene balls as an intermediate step. Many useful unit steps in the base-line fabrication process were optimized. We conclude that the P3HT in chlorobenzene does not form a good film over aluminium gate due to high surface energy and hence different methods and solvents are tried. It is found that a mixture of chlorobenzene and xylene as the solvent for P3HT can serve the purpose. The film quality of P3HT obtained on Al still results in pin holes and crevices, and hence the active area of the device is kept small i.e. 0.5mm2 to avoid possibility of such pin holes in test devices. The output characteristics showed that with applied drain-source voltage of 1 V, the device is capable of providing current close to hundred micro-ampere with moderate gate control, which needs to be improved upon with better processing strategies. The drain source current shows evidence of onset of space charge limited current, whose onset is controlled by gate voltage.

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Title : Organic Solar Cells:Impact of Photogenerated Current Variation on Fill Factor, Series & Shunt Resistances

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Abstract

Unlike inorganic solar cells, light absorption leads to creation of bound electron hole pairs in organic solar cells, whose dissociation into free carriers depends on electric field. Hence, the photo generated current in organic solar cells has voltage dependence. Investigating the effect of this voltage dependence in determining the cell parameters like fill factor and extraction of circuit model parameters like series and shunt resistances will give more insights into working of solar cell. In this work, firstly, an analysis of fill factor in organic solar cells is described which highlights the importance of variation of photo-generated current with applied voltage. It is shown that a ratio of short circuit current and dark current at open circuit voltage close to unity together with open circuit voltage smaller than built-in voltage is required to obtain a high fill factor. Two new parameters are proposed to quantify the relative impact of variation of dark and light generated current with voltage. Simulation results for bulk heterojunction solar cells are used to illustrate the factors affecting fill factor in organic solar cells. Secondly, series and shunt resistances and their light intensity dependence have been investigated. These parasitic resistances are commonly found to have different values under dark and light conditions. One contributing factor for this is neglect of dependence of photogenerated current on voltage in the resistance extraction process. The second reason for the observed discrepancy between light and dark resistances is neglect of partitioning of series resistance into intrinsic and extrinsic components and their appropriate placement in the equivalent circuit model of the solar cell. Simulation results obtained for bulk heterojunction solar cells show that discrepancy between light and dark resistances disappear when the two effects mentioned above are properly accounted for.

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Title	**:**	***Automatic Transistor Sizing using Ant Colony Optimization Algorithm***
Author(s)	**:**	***Gupta Himanshu***
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Supervisor(s)	**:**	***Ghosh Bahniman***

Abstract

Transistor size optimization is an important aspect of circuit designing. Small and non-complex circuits can be designed easily by doing some manual calculations and circuit simulations. But, with increasing complexity of circuits, this work becomes difficult and requires a lot of time. Therefore, tools and techniques for automatic transistor sizing are of great importance, in the area of circuit design. The goal of this thesis is to present Ant Colony Optimization (ACO) algorithm, as a tool to find transistor sizes in digital and analog circuits, for given output specifications. ACO is a swarm intelligence algorithm, which was first proposed in early nineties, to solve the problems of combinatorial optimization. In this thesis, a modified version of this algorithm, (for continuous domain) has been implemented in Perl, to use it for automatic transistor sizing. The whole work can be divided in three parts, based on the type of circuits optimized. First part involves optimization of four digital circuits, of different complexity. These circuits are optimized using ACO, to obtain optimum balance between delay and power. In the second part of thesis, widths of transistors are found for three analog circuits, to achieve the given specifications. Finally, ACO has been tested on an Analog to Digital (ADC) Converter. Performance of the ADC is improved by optimizing the widths of the transistors, to minimize the error in the output of the ADC. In the past, one of the most popular evolutionary algorithms, Genetic Algorithm (GA) has been found to be effective in optimizing transistor sizes. Therefore, to examine the solutions achieved by ACO, all the circuits are optimized by GA also. Results show that ACO is better than GA in finding transistor sizes. Also, ACO takes less time in optimization process. In this work, Perl code of algorithms has been coupled with HSPICE to do circuit simulations. All circuits are simulated using BSIM3v3 MOSFET models in 0.13µm, 0.18µm or 0.35µm CMOS processes.

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Title : *Spin Transport in Nanowires*
Author(s) : *Kumar Ashish*
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Abstract

The research in the area of spintronics is fast gaining momentum due to the promise such spintronics based devices have shown. It involves utilizing the spin degree of freedom and this can provide plethora of advantages over conventional electronics by providing new capabilities and new functionalities. Spintronics with semiconductors is very attractive as it exploits both the properties of an electron- charge and spin. It can combine the capabilities of semiconductors with the capabilties of the magnetic materials. Such devices can possess gather storage, logic and communication capabilities on the same chip thus replacing several components. We simulate spin relaxation in semiconductors which is a critical parameter in determining the suitability of a material for spintronics. We study spin transport by coupling semiclassical Monte Carlo method with spin density matrix calculations. Spin dephasing is caused by D'yakonov-Perel' relaxation which is caused due to bulk inversion asymmetry (Dresselhaus spin-orbit coupling) and structural inversion asymmetry (Rashba spin-orbit coupling). Spin flip due to Elliott-Yafet is also taken into account. Spin relaxation is investigated in silicon, germanium, SiGe, InP and InSb for different dimensionality systems. The effect of temperature and driving electric field on spin relaxation length is studied. The spin relaxation length increases with confinement of motion and with decrease in temperature. The relaxation rate is also found to depend on the initial polarization of spin.

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Title : *Device and Circuit Performance Evaluation and Improvement of Tunnel FETs*

Author(s) : *Mishra Rahul*
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Abstract

With the reduction in device size from micro scale to nano scale the present day devices, ie the MOSFETs are facing issues like gate overdrive, high subthreshold swing, and other short channel effects. These drawbacks counteract the advantage that we expect from scaling of devices. Recently work has been done on tunnel FETs which are low power device and their performance is not affected by device scaling. Extensive simulation and little experimental work has been done on tunnel FETs, and constant efforts are made to improve their performance in terms of high on current, high Ion – Ioff ratio, low subthreshold swing. For achieving the above various modifications have already been suggested in the basic silicon TFET. Our thesis aims at further improving the performance of tunnel FETs, analyzing their circuit performance and also finding new ways of modeling TFETs. In first part of our thesis we have worked on device simulation of SiGe TFET with SiGe on source side, we performed circuit simulation of this device with already available device simulation software, and we have also suggested a way to improve this device. In the second part we have simulated extended channel SiGe TFETs, looked at its characteristics. We modeled this TFET is Pspice and performed its circuit simulations. In the last part we look at the NEGF method to model nano scale devices in our case TFETs. We have performed this modeling on complete SiGE and GaAs TFETs.

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Title ***: Design and Analysis of Low Power SRAM***
Author(s) ***: Nandal Vikas***
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Supervisor(s) ***: Ghosh Bahniman***

Abstract

The aim of the thesis is to obtain a low leakage 8T SRAM cell while keeping hold static noise margin during read operation. 8T SRAM cell uses the concept of stacking to reduce sub-threshold leakage in the standby mode. The 8T SRAM cell provides 58.36% reduction of leakage power in the hold mode while storing 0. As most of the bits in the instruction and data caches are 0, therefore emphasis is to reduce the sub-threshold leakage current when cell stores 0.Due to the voltage scaling read SNM degrades so another stacking in 8T SRAM cell is used to isolate storage nodes from bit-lines. This proposed 8T SRAM cell improves read SNM by 127% which is comparable to the hold SNM. Cell can be operated under low voltage environment condition due to SNM free read operation. Read SNM of 8T drastically improved from 220mv to 500mV whereas hold SNM degrades by 50 mV and comes out to be 670mV. Write margin is also improved by -70mV therefore proposed topology is easy to write as compared to 6T. The performance of the cell with 4k memory using 8 to 256 decoder and column multiplexer is done. Memory access time for read 0 and 1 is 1.081ns and 1.066ns, respectively whereas for write 0 and write 1 is .818ns and 2.348ns, respectively. This difference in write time is due to the use of stacking concept to reduce sub-threshold leakage current. All designs have been performed in 180nm CMOS technology from Cadence EDA tool suit.

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Title **:** ***Implementation of High Speed and Low Power ADC***
Author(s) **:** ***Navhakar Rahul Shivaji***
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Supervisor(s) **:** ***Bahniman Ghosh***

Abstract

In cutting edge high-speed CMOS technology, information transmitted in sub nanosecond pulses. As the CMOS technology continue to scale down, signal processing is favourably done in digital domain, which requires Analog to Digital converters (ADCs) to be integrated on-chip. A high speed, high resolution ADC is very difficult to design, it consumes more power too. Among various ADC architectures, the two step Flash ADC architecture is best suited for low power and 6-bit resolution. To decrease the area, power consumption, and cost while maintaining 6-bit accuracy, the architecture is divided into course Flash ADC and Fine Flash ADC connected through current steering Digital to Analog convertor (DAC) and residue amplifier. In this thesis, 6-bit two step Flash ADC operated at the 2 GS/s is designed, with optimized performance for frequency domain applications. To get the best performance, the coarse convertor of 2-bit and the fine convertor of 4-bit are chosen. The comparator design is done with optimum power and area with two power supplies. Intermediate state accuracy increased by thermometer coded current steering DAC. The DNL and INL specifications are chosen to be 0.8 LSB and 0.9 LSB respectively. For a sampling frequency of 2 GS/s, the SFDR is greater than 47dbc for the signal from dc to 70 MHz and the SNR is greater than 34dB. The power consumption is only 9.35mW, and figure of merit equals to 0.146 (pJ/step) at the power supplies of 1.8V and 0.75V. The key building blocks like sample and hold circuit, comparator, residue amplifier, current steering DAC are designed. Layout and simulation of functional element at the block level is performed. All the designs are done in 180nm CMOS technology from UMC (United Microelectronics Corporation) Cadence EDA (Electronic Design Automation) tool suite.

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Title : ***Effects of Device Geometry on Performance of P3HT:PCBM Organic Solar Cells***

Author(s) : ***Agarwal Vishal***
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Supervisor(s) : ***Iyer S Sundar Kumar***

Abstract

Finite resistivity of ITO is one of the limiting factors in designing large area organic solar cells. The geometry of ITO anode in the solar cell significantly affects the performance of these devices. Till now, most of the research is focused in the area of increasing efficiency and lifetime, but as we are approaching commercially viable limit of OSCs, we will have to take into account the effects of device layout and geometry. This work focuses on design of various solar cells having different geometrical parameters such as area, perimeter and aspect ratio. The effect of each parameter is determined with the help of devices fabricated with two different designs. First part of this thesis describes an empirical model for OSC to study effect of series and shunt resistance. It is shown that efficiency and fill factor decay exponentially with increasing series and decreasing shunt resistances. A numerical simulation for efficiency with geometry is also carried out. Using the numerical simulations, it is shown that there is a 'critical width' beyond which efficiency degrades drastically. While designing large area solar cells, this width should not be surpassed. In the second part of thesis, two designs with different device layouts are proposed. First layout has variations in area, perimeter and aspect ratio. Second design has variations in terms of only perimeter and aspect ratio and the area is kept constant in all devices. Using the first design, it is proved that efficiency decays with area and the series resistance increases linearly with area. Using regression analysis, role of each parameter is determined. It is shown that for the design under consideration, area and perimeter are most critical parameters. Aspect ratio plays less role in this case but poor aspect ratio can start dominating performance by contributing to large series resistance. Using second experiment, it is confirmed that efficiency degrades with increasing ITO width even when area is constant.

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Title	***:***	***Optimization Of Deposition Parameters Of Aluminium Doped Zinc Oxide For Organic Solar Cell Applications***
Author(s)	***:***	***Jannu Sandeep Kumar***
Roll No	***:***	***Y9104062***
Supervisor(s)	***:***	***Iyer S Sundar Kumar***

Abstract

In the past decade, organic solar cell technology is growing at a high pace because of low cost of production involved and is environmental friendly. Its basic structure consists of an organic semiconductor sandwiched between two metal electrodes of different work functions. Of these two electrodes, one is a transparent electrode, which is typically anode, and is called transparent conductive oxide (TCO) to pass the incident light. Indium tin oxide (ITO) is widely used TCO but due to scarcity of indium, other suitable alternatives are being studied. This thesis mainly focuses aluminium doped zinc oxide (ZnO:Al) as a suitable alternative TCO and solar cell is fabricated on it to study its performance. ZnO:Al thin film is deposited on the glass substrate by the process of sputtering. The characteristics of the deposited film vary widely with deposition variables. Design of experiment is carried out to study the combined effect to deposition pressure and deposition temperature on the properties of sputtering ZnO:Al. The best value of resistivity that is obtained is 2.29x10-4 Ω-m. The effect of hydrogenation on the electrical properties of the zinc oxide film is also studies and the hydrogen flow rate is optimized for best conductivity film. The best possible resistivity obtained is 1.01x10-4 Ω-m. P3HT:PCBM solar cell is fabricated on the zinc oxide sputtered glass substrates and on the commercially obtained glass substrates for comparision. The efficiency of the zinc oxide based solar cell obtained is less when compared to ITO based solar cell which is understood as sheet resistance of ITO is very much low when compared to ZnO:Al.

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Title : ***Extraction Of Mobility From Low Voltage Characteristic Of Single Layer Single Carrier Organic Semiconductor Device***

Author(s) : ***Biswas Hiranmay***
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Supervisor(s) : ***Mazhari Baquer***

Abstract

Organic electronics has attracted significant interest in the past decades, because of the possibility of affordable and user friendly products based on flexible substrates that are less damaging to the environment. While organic light emitting diodes (OLEDs) have already reached mass production level, other devices like organic solar cells (OSCs), organic field effect transistors (OFETs), organic rectifying diodes and organic memories are still undergoing extensive fundamental research. All conceivable applications have in common that their functionality within a device is strongly determined by the charge carrier transport properties, i.e. the carrier mobility. As a result, understanding of transport mechanisms, measurement of mobility and techniques for its improvement are very important for eventual success of organic electronics.The Electron and holes mobilities are very low in organic semiconductors as compared to inorganic semiconductors. The mobilities cannot usually be measured by conventional measurement techniques that use Hall effect Although a number of measurement techniques have been proposed, the simplest and perhaps most widely used technique uses space charge limited current in a single layer single carrier device to estimate mobility. This technique assumes that current varies quadratically with applied voltage which however is many times not observed in experimental data. The present work describes the limitations of SCLC technique and proposes a correction that improves its accuracy. In addition, the low voltage ohmic characteristic is proposed as an alternative operating region from which mobility can be extracted as well. Two dimensional simulation results show that the proposed technique is less sensitive to errors resulting from higher injection barrier heights, small residual built-in voltages due to differences in electrode workfunctions and field dependent mobility.Single layer hole only devices with ITO/PEDOT as one electrode and Gold as the second electrode with three different organic materials were fabricated and characterized. Mobility was extracted using conventional SCLC, corrected SCLC and the proposed low voltage technique. For both Pentaceneand P3HT:PCBM blend, a clear SCLC region with a power exponent of 2 was not observed even after correction thereby precluding accurate measurement of mobility from these techniques. However, a well defined ohmic region was observed allowing mobility to be estimated from the low voltage characteristics. For P3HT device, the corrected SCLC showed an exponent close to 2 and mobility estimated from it matched closely with that estimated from low voltage characteristics. These results show that low voltage characteristics in single layer devices can be better than conventional SCLC technique for estimation of mobility.

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Title	***:***	***A Threshold Voltage and Mobility Compensation Circuit for AMOLED Displays using Time Ratio Gray Scale***
Author(s)	***:***	***Pandey Vishal***
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Abstract

Abstract Research in Active Matrix Organic Light Emitting Diodes (AMOLED) displays is being actively pursued to develop high resolution flat panel displays which can cater to the needs of high definition media formats. Design of backplane using Low Temperature Poly Silicon (LTPS) Thin Film Transistors (TFT) is particularly challenging for AMOLED displays. A primary requirement for the pixels of these displays is that they must be insensitive to the variation of various device parameters like threshold voltage (Vth), mobility of charge carriers (μ), device degradation etc. These factors bring non-idealities in the display, causing its overall performance to degrade with time. Although many techniques have been developed, most of them focus on threshold voltage compensation and cannot address mobility variation suitably without putting excessive complexity in design. This work investigates one of the recently developed method to compensate both threshold voltage and mobility variations while keeping minimum complexity in design, with primary focus on LTPS technology. The method used here consists of a 2-TFT and 1 capacitor circuit utilizing the local matching property of Poly-Si technology and a negative feedback for both threshold voltage and mobility compensation. This circuit is able to perform well in a range of 0-1.84 μA with resulting non-uniformity being less than 5% for both parameters (Vth and μ) at several different current values. The total programming time is found to be around 0.5 μs which makes it optimum for high resolution displays. A modified Time Ratio Grayscale (TRG) method is proposed and is demonstrated by generating a 64 level grayscale in 0-0.57 μA range with non-uniformity in threshold voltage being less than 6%. 256 level grayscale is obtained using modified TRG method in the extended current range of 0-1.4 μA with non-uniformity less than 1% for Vth and μ variations.

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Title : *Studies On III-V Tunnel TETs*
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Supervisor(s) : *Ghosh Bahniman*

Abstract

In recent years, MOSFETs have high subthreshold swing (S) and leakage current (IOFF) because of continuous scaling. This has the effect of static power increasing in electronic systems. To improve the energy efficiency of electronic systems, alternative devices which have low leakage current and low swing S are investigated to replace the MOSFETs. Tunnel FETs become attractive for low power applications because of low leakage currents and exhibit low subthreshold swing S. Basically, tunnel FET (TFET) is a p-i-n diode with gates over the channel region and works on the principle of band to band tunneling. All simulations are carried out in Silvaco Atlas. A non local band to band tunneling model is used for tunneling. Various hetero junction Silicon (Si) tunnel FETs are investigated which are suitable for low power applications. Source of hetero junction Si TFETs are made up of different III-V group materials like InGaSb, InAs and InP and IV group materials like Si and SiGe. Different channel lengths of tunnel FET are analyzed. Tunnel FETs are modeled in model editor. After modeling, circuits are designed in Orcad Capture. Supply voltage is 0.5 V and 1 pF is used as a load for all circuits. Performance of these circuits is evaluated in terms of average delay, dynamic power and leakage power. To improve the ON currents, total body tunnel FETs are used. In total body tunnel FETs, source, drain and channel are made up of same material. Various group III-V and group IV materials are used in total body tunnel FETs. Effects of channel length variations are analyzed. Digital circuits are implemented using tunnel FET models and circuit performance is evaluated. A double gate MOSFET is modeled whose structure is similar to TFET structure and its circuit performance is compared with tunnel circuits.

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Title **:** ***Design of high frequency CDBA filters***
Author(s) **:** ***Sathish Babu Kotaru***
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Supervisor(s) **:** ***Biswas Animesh***

Abstract

Recently in the analog and RFIC design the support of the CMOS processes continues to mature. These processes are fastly becoming the choice of technology in RFIC development. This transition is invoked due to the fact that CMOS processes are less expensive and more conductive to large –scale integration. With the advent of mobile communication systems and portable electronics the importance of low voltage circuit design has been highlighted. For low voltage operation current mode circuits are extremely useful. These current mode circuits are receiving tremendous interest in the analog signal processing applications as an alternative to voltage-mode circuits. An active element which can operate in both voltage and current mode circuits named by current differencing buffered amplifier (CDBA) and is free from input parasitic capacitances is found to be appropriate for performing high frequency operations. This CDBA consumes a low power of about 1 mW is used to realize the RF filters. In this work by using the above mentioned CDBA a high quality band pass filter suitable for GSM cellular base station has been designed with a center frequency at 890MHz.This active band pass filter consumes a power of 10 mW and it provides a gain of 14.7 dB at its center frequency. A notch filter suitable for low-Q applications has been designed at a center frequency of 1.7MHz with this CDBA. For simulations purpose cadence UMC 180 nm processes technology has been used.

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Title **:** ***Modified Annular Slot Antenna loaded with Meander Slot***
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Supervisor(s) **:** ***Harish A R***

Abstract

In this thesis the research undertaken is in the area of compact RF/microwave antenna design which may be used for RFID applications. The design involves radiation on both sides of the substrate and thus slot antenna is the best possible solution. The thesis also involves the design of a compact annular slot antenna with broad bandwidth. A study of several compact slotline antennas reported in the past has been carried out and they were mainly designed for dual-multi bands for WLAN. Here the approach taken was to introduce meander slot within the annular slot to achieve another resonance near to annular slot resonant frequency and to bring these resonances closer to increase bandwidth. The analyses of annular slot and meander slot were carried out separately and equivalent circuit based on transmission line model was proposed for both the antennas. After analyzing two antennas independently they were combined them into single structure. The combined structure had a much large bandwidth compared to the individual elements. An equivalent circuit was proposed for the annular slot antenna loaded with meander slot. Circular polarization is achieved by incorporating crossed meander slots within the annular slot antenna, and by dual-feed. The dual-feed to the structure is based on hybrid coupler designed on the same substrate as of the antenna. The size of the proposed antenna is small 0.378λ x 0.378λ and has bandwidth of approximately 30%. The antenna has a gain of approximately 3.5 dBi. The simulated and measured results are in good agreement. The small size of the antenna makes it very suitable for use in modern RF/microwave wireless systems which require compact, low cost, and high performance circuits. Moreover, its circular polarization makes it more attractive for various applications other than RFID systems.

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Title ***:*** ***Wide Band Printed Dipole Antenna***
Author(s) ***:*** ***Behera Amiya Ranjan***
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Abstract

Nowadays, different telecommunication protocols, systems, and networks have born in order to satisfy the requirements of recent wireless technologies. Also, thanks to modem electronics and the struggle for a growing market share, more and more advanced multipurpose handheld gadgets like smart phones, Personal Digital Assistants (PDA), iPad etc. have evolved rapidly during the last two decades. These devices need to communicate through different protocols, such as Bluetooth, Wi-Fi, wireless local loops (WLLs) based on Fixed Wireless Access (FWA) band, or ISM (Industrial, Scientific and Medical) band etc., by using only a single antenna. Hence, there is a rising need for wide band antennas which are of small size, low profile, light weight, low cost as well as easy to fabricate and install. Printed dipole antennas, being omnidirectional in nature, are very suitable for such wireless application. But, they are limited by their inherent property of narrow bandwidth. In this work, we present a novel technique to improve the input bandwidth of conventional printed dipole antennas without actually increasing their size. We show that the proposed modification results in a considerable increment (19% to 67%) in the input bandwidth. We also propose an equivalent circuit model for this 'flared-arms' printed dipole configuration which can be readily used to predict the behavior of similar structures. A complete procedure has been described to arrive at the values of these circuit elements. Finally, the performance of the antenna is demonstrated by a fabricated model and the comparisons of experimental results with simulation.

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Title **:** ***Performance Enhancement of RFID Tag***
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Abstract

In RFID system, one of the simplest ways of attaching the RFID tag to an object is to simply attach it on surface of the object. It is well known that the performance of the tag is strongly in uenced by the properties of the object on which it is attached. Several techniques have been proposed to minimize the interference and maximize the performance of the tag. However, no attempts have been made to modify the constitution of the object itself to improve the tag performance. For reliable operation, a tag needs to be protected from the environment. There are several applications that demand that the placement of the tag should not hinder the performance of the object being tagged. e.g. handle(grip) of a tool. This work relates to placing the tag inside an object. The performance of a tag degrades when it is placed inside an object. We have suggested a technique to improve the performance by making some changes to the properties of the material. The changes could be in the form of introducing an air pocket, another material having slightly different electrical properties, or modifying the geometry of the space holding the tag. The work done includes demonstration of performance enhancement for cylindrical, cuboid and planar (sheet type) dielectric objects using simulations. The performance enhancement is also demonstrated experimentally for cylindrical object by measuring the threshold power and the experimental observations are compared with the simulation results. The technique for tag performance enhancement has been tried on a screwdriver using simulations and experimental measurements. A detailed parametric study is also carried out using simulations to show the effect of various parameters like tool length and position of metallic inclusion on performance of tag.

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Title : *New 1.8v current feedback operational amplifier*
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Supervisor(s) : *Biswas Animesh*

Abstract

Now, electronic circuit design, most Voltage Op-Amp (VOA) is replaced by Current Feedback Op-Amp (CFOA) in high frequency application. With better performance and low cost. Some designers are not intimidated by Current Feedback Op-Amp (CFOA), thus they don't take advantage of the CFOA's superior frequency performance. CFOA have become pervasive because they have an architectural advantage that delivers high bandwidth and very high slew rate at low supply current application. In this thesis work, we present a novel CMOS low-voltage current feedback operational amplifier (CFOA) . The proposed CFOA based on a new positive second-generation current conveyor (CCII+). The proposed circuit allows almost zero input and output voltage and current tracking error. Also it reduces the offset voltage and has a very low inverting input resistance and provides high driving current capabilities. The CFOA is operating at supply voltages of ±0.9 V with total stand by power dissipation of .5mW.The circuit exhibits better than 10 MHz bandwidth and ±4.6mA current drive capability. Cadence virtuoso simulation results are given using 0.18μm UMC technology for the proposed CFOA.

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Title : *An Inverse Scattering Procedure to Design Microwave Filters*
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Supervisor(s) : *Akhtar M Jaleel*

Abstract

The planar microwave fi lters have gained much popularity in recent years for various RF and microwave applications because of their advantages such as light weight, low cost, their easy integration with active devices etc. A number of methods, both analytical as well as numerical, have been proposed in the past to design the microstrip and other types of planar lters. The full 3-D electromagnetic eld simulators are nowadays quite often being used for the design of these lters because of the available computational power, and the strong graphical user interface provided by some of the commercially available simulators. A new analytical approach for the design of microwave planar lters is presented. The method is based on the solution of an inverse scattering problem, and involves the reconstruction of the impedance pro le corresponding to the dimensions of the lter in terms of its speci ed frequency response. The overall procedure requires the re-formulation of the famous Riccati di erential equation in terms of the spectral domain re ection coe cient data and the impedance pro le. Then an algorithm is developed to generate the impedance pro le of the microstrip lter as a function of space. The frequency domain re ection coe cient data are then used in the proposed algorithm to generate the continuously varying impedance pro le of the nonuniform microstrip line that can generate the desired lter response. The continuously varying impedance pro le is nally transformed into the geometry of the microstrip lter using the commonly available empirical relationships. The proposed method is very general and it can be applied to reconstruct any type of impedance pro e as a function of distance. The applicability of the overall procedure is veri ed by designing a number of microstrip lters of various types. The design is independently veri ed using a full 3-D electromagnetic eld simulator and these lters are fabricated based on the frozen designs. First of all, the theory of inverse scattering has been applied to for impedance reconstruction. A number of generalized continuous and discontinuous impedance pro les have been presented. Afterwards, the proposed approach is used to design a number of continuously varying microstrip lines. Finally, the proposed method is applied for the design of planar microwave lters. For this purpose, the microstrip lters of various types are designed using the proposed technique, and the results are compared with other available methods.

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Title **:** ***A Compact Zeroth Order Resonating Antenna Design Using Mushroom Type Of Structure With CSRR***

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Supervisor(s) **:** ***Srivastava Kumar Vaibhav***

Abstract

In the last few years the wireless communication systems have grown tremendously, so to meet the requirement of the situation, great efforts are being made in the field of antenna design especially related to its size reduction. Evolution of the metamaterial theory especially related to composite right left handed (CRLH) transmission line gives the new dimension to realize the compact microwave circuit in the planer environment. In this dissertation we are presenting the compact zeroth order resonating (ZOR) antenna designs based on the conventional mushroom type of structure (CMT) and complementary split ring resonator (CSRR). Along with this, two port lumped circuit modeling of the proposed ZOR antenna's constituent unit cell is also included. The array of mushroom type structure is basically CRLH transmission line. ZOR is the unusual property of the open or short circuit terminated CRLH transmission line, which allows the resonance in the transmission line even at the zero propagation constant. In the published literature, we have seen that the mushroom type structure based ZOR antennas resonance frequency is determine by the lumped component of the shunt arm of the unit cell. Inclusion of CSRR in the CMT structure adds the lumped components in the shunt arm and reduces the value of the shunt resonance frequency and thus a more compact ZOR antenna can be realized. In this work, three designs of ZOR antenna have been proposed. The first two designs are 4 cells ZOR antenna and epsilon negative (ENG) ZOR antenna. The third design is the improved radiation efficiency model of the first design. The performance of the first and third design has been demonstrated by the comparison between the simulated and the fabricated results.

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Title	***:***	***Design Of Stair-Shaped Printed Wideband Fractal Antenna Using Defected Ground Structure***
Author(s)	***:***	***Shrivastava Abhay Kumar***
Roll No	***:***	***Y9104002***
Supervisor(s)	***:***	***Srivastava Kumar Vaibhav***

Abstract

There is a rapid growth in area of wideband and Ultra wideband technologies such as wireless communication system. The Wi-Max, W-LAN, Wi-Fi, GPS etc. services all needs compact, small size, broadband antenna. The required antenna must have pattern stability and stable gain over the operating band. Various geometrical shapes, derived from fractal mathematics have been investigated. Fractal shapes means self similar, scaled dimensions. Fractal antennas based on Sierpinski Gasket and Minkowski Island iteration, were simulated to investigate, its multiband, or wideband features. Applying the concept of fractal geometry and defected ground structure, the physical size of antenna can be reduced effectively. In proposed design, Antenna structure has been simulated in order to increase the bandwidth, and size reduction of overall structure. The field radiation pattern and gain pattern, cross and co-polarization have been investigated in principle E-plane and H-plane. Uniformity and stability of pattern have been achieved over the entire frequency band. The proposed “Stair Shaped Fractal Printed Wideband Antenna” of size 20mm x 45mm (including substrate area) was developed, which is the reduction of six fold size of existing conventional Microstrip antenna.

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Title : *CPW-fed Wideband Slot Antennas*
Author(s) : *Meena Rajesh*
Roll No : *Y5827357*
Supervisor(s) : *Harish A R*

Abstract

As many wireless applications are evolving rapidly in the recent times, antennas with a wide bandwidth play an essential role. To achieve the high bandwidth with the limit on the size & cost, it becomes a challenging task. In this thesis, we use the technique of enhancing the bandwidth of the printed slot antennas using parasitic patches. Parasitic elements can be used in the printed slot antennas by placing the elements inside the slots. Depending on the slot shape and size a suitable parasitic patches can be used so that one can achieve a wide impedance bandwidth without actually increasing the size of the antenna. In this work, two new modi ed antenna structures are presented based on CPW-fed tapered-elliptical slot antenna. Both the antennas show wide- band performance. The previously reported CPW-fed tapered-elliptical slot antenna gives bandwidth upto 42%. So rst, the structure is modi- ed by introducing triangle shaped parasitic elements inside the slot re- gion such that depending on the size and location of the parasitic elements coupling can be controlled. Another resonance can be brought closer to the fundamental one by optimizing the parameters to get desired result of wide bandwidth of 73%. Another modi cation has been done in the proposed parasitically loaded CPW-fed tapered-elliptical slot antenna by introducing another resonance by cutting a triangular slot in the tapered region between the pair of slots. With the second proposed design a over- all wide impedance bandwidth of 93% is achieved. A detailed parametric study for each of the three designs has been presented to show the e ect of relevant geometrical parameters on the performance of the antenna. Both modi ed designs are fabricated on FR4 substrate and measurements show good agreement with the simulation results.

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Title	***:***	***Group Delay Based Methods for Robust Speech Source Localization using Shrinkage Estimator***
Author(s)	***:***	***Mandala Rohan***
Roll No	***:***	***Y6927239***
Supervisor(s)	***:***	***Hegde Rajesh Mahanand***

Abstract

The group delay function has been used conventionally in temporal spectral analysis and feature extraction for speech recognition. Recent work on the MUSIC-Group Delay spectrum has effectively used this technique for direction of arrival estimation and localization of multiple speech sources. In this thesis a new approach to robust speech source localization using MUSIC-Group Delay spectrum using a shrinkage estimator is proposed. This method is able to resolve spatially close speech sources using a minimal number of microphones. Subsequent applications in hands free distant speech recognition is also proposed in this thesis. The MUSIC-Group Delay spectrum utilizes the differential phase of the MUSIC spectrum. However it follows the conventional method of an eigen value decomposition of the correlation matrix followed by searching for the orthogonality of the noise sub space with the steering vector. A better estimate of the covariance matrix can obtained using the method of shrinkage estimation. This estimation is more flexible and robust as illustrated in the detailed analysis carried out in the thesis. The proposed method exhibits robustness to sensor perturbation errors and in reverberant room environments as illustrated by the detailed experimental analysis. Additional analysis is also performed using the Pisarenko-Group Delay spectrum in terms of real time performance. Studies on the effects that hardware inaccuracies have on DOA estimation are also carried out. The additive property of the MUSIC Group Delay spectrum is illustrated via the root-MUSIC polynomial analysis. A minimum phase analysis of this high resolution method of DOA estimation is also presented with relevant experimental results. Average error deviation plots have been presented to compare the performance of different methods in both reverberant and non reverberant conditions. A Cramer-Rao lower bound (CRLB) analysis is performed and the performance of the proposed method is compared to conventional methods of DOA estimation with respect to the CRLB. The application of the proposed method in hands free distant speech recognition is studied. Experiments on distant speech recognition in clean and reverberant conditions carried out by training filter and sum beamformers on the S-TIMIT and the MONC data indicate reasonable improvements over conventional time delay and sub space based methods.

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Title : *Loop Free Multipath Routing Algorithm*
Author(s) : *Singh Rashmi*
Roll No : *Y6927378*
Supervisor(s) : *SinghYatindra Nath*

Abstract

Single path routing that is currently used in the internet routers is easy to implement as it simplifies the routing tables and packet flow paths. However it is not optimal and has shortcomings in utilizing the network resources optimally, load balancing & fast recovery in case of faults (fault tolerance). The given algorithm resolves all these problems by using all possible multiple paths for transfer of information, while retaining loop-free property. We have proposed a new dynamic loop-free multipath routing algorithm which improves network throughput and network resource utilization, reduces average transmission delay, and is not affected by faults in the links and router nodes. The main idea of this algorithm is to maintain multiple possible next hops for a destination along with weights. At every node, the traffic to a destination is split among multiple next hops in proportion to the estimated weights. The number of multiple next hops also changes depending on the traffic conditions, but it is never less than one.

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Title	***:***	***Group Delay-Cross Correlation Methods for Single Channel Speaker Segregation***
Author(s)	***:***	***Pandit Pranav***
Roll No	***:***	***Y6927327***
Supervisor(s)	***:***	***Hegde Rajesh Mahanand***

Abstract

Single channel speaker segregation is a very challenging task. The presence of noise and reverberation can be handled by different signal processing techniques given that the nature of these interferences is different from that of the speech signal. However, when a competing human interference is present within the same channel the acoustic characteristics of speech from the competing human speaker is very similar in nature to the desired speech signal, which makes the single channel speaker segregation problem an unsolved issue in speech research. In this thesis, a novel method of single channel speaker segregation using the sub-band group delay-cross correlation function is proposed. Techniques like the sub-band instantaneous frequency correlation have already been used to a limited extent for speaker segregation. However, the proposed method is based on the group delay of frequency sub-bands. The group delay function, which is defined as the negative derivative of the phase spectrum yields robust spectral estimates due to its inherent mathematical characteristics. The group delay spectral estimates are computed over sub-bands after the speech signal is passed through a bank of filters, based on a multi-pitch algorithm, to obtain the output of each frequency band. A correlation matrix is then computed from the group delay spectral estimates of each sub band, which represents the correlations between the various sub-bands of the mixed speech signal. The grouping of the sources (harmonics) present in the mixed speech signal is carried out by using the graph cut method. An iterative graph cut method is also proposed to further improve the performance of the grouping. The signals are reconstructed from the respective groups as obtained from the iterative graph cut method. Spectrographic masks are then estimated and applied on the reconstructed signals to further improve the quality of the separated signals. Experiments are conducted to evaluate the performance of the proposed separation method using several objective and subjective evaluation criteria. In order to check the efficacy of the method in practical applications, experiments on multi speaker speech recognition are conducted using mixed speech data from the GRID corpus. The proposed method demonstrates reasonable improvements in terms of the evaluation criteria used and the speech recognition performance when compared to other conventional methods detailed in this thesis

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Title	***:***	***3d Generation, Compression And Processing Of Novel Views From Light Field Data***
Author(s)	***:***	***Popli shivaranjani***
Roll No	***:***	***Y9104071***
Supervisor(s)	***:***	***Venkatesh K S***

Abstract

Light-fields are 4-dimensional databases representing the value of light rays permeating a scene from different positions and directions. This enormous information can be easily and effectively deployed for image-based rendering and visual analysis. Thus, it can render objects with a high degree of realism without having to depend on scene complexity. This work reproduces some results based on light-field data such as novel-view synthesis, variable focus and variable aperture. It advances into compression of light-fields as an extension of novel-view generation and introduces the concept of "true zoom" in terms of camera motion behind and beyond the camera plane. It also does some analysis on the effect of noise and distortion on light-field techniques, which in effect suggest that light-field synthesis and noise/distortion addition commute with each other.

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Title **:** ***Robust Automatic Image Annotation of Large Image Databases***
Author(s) **:** ***Mishra Anand***
Roll No **:** ***Y6927062***
Supervisor(s) **:** ***Venkatesh K S& Karnick Harish***

Abstract

The explosive growth of web and increased public reach of digital cameras whether handheld or of cell phones has resulted in large collections of images on the internet. This huge collection of images is poorly organized and indexed. Famous search engines like Google use the metadata associated with the images for their search queries instead of the image content itself. A large portion of the images on the internet is not reachable by search engines as either do not have proper metadata associated with them or have no metadata at all. Thus associating metadata or automatically annotating images based on their content is a very crucial step if we want to have content based search using query words. The problem of annotating or tagging images based on their content in large databases is a classical unsolved problem in Computer Vision. Despite considerable research done in this area providing quality annotation on large databases remains a big issue. Manual annotation of these images can be very expensive and almost impossible since the number of images is in billions and is increasing. Also retrieving images based on similarity search between a visual query and a database image (CBIR) is not very effective as it does not lend itself to textual queries. In this work we propose a new framework for automatic image annotation. It uses a weighted combination of different feature-classifier pairs to tag a particular image which makes the algorithm more robust. The weights are selected depending on the query image at hand by projecting it onto the subspaces correctly classified by the feature-classifier pairs on the training set. The feature-classifier pairs used are also in turn selected depending on the training dataset images which make the framework more logical and consistent.

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Title **:** ***Detection of Frame Erasure for Turbo Coded Offset QPSK Signal Over Quasi Static Fading Channel***

Author(s) **:** ***Sharma Ekant***
Roll No **:** ***Y9104015***
Supervisor(s) **:** ***VasudevanKasturi***

Abstract

One of the common problems associated with wireless communication is fading. In this thesis, we address the problem of detecting turbo coded offset QPSK signals, transmitted over quasi-static flat fading channels. Here the channel gain is assumed to be constant over one frame of data and varies randomly from frame to frame. The channel gain is assumed to be a zero-mean Gaussian random variable. When the magnitude of the channel gain is close to zero, the frame gets severely attenuated and has to be discarded (erased), since the data cannot be recovered from such frames. The topic of this thesis is to detect such frame erasures. The procedure for frame erasure is based on a differential correlation (DC) algorithm. The input to the DC algorithm is the received signal. The average and the peak power of the output of the DC algorithm is computed. If the average and ratio of the maximum to the average power exceed a certain thresholds, the frame is declared as valid, else it needs to be erased. Besides fading and additive white Gaussian noise (AWGN), the other impairments considered in this thesis are carrier frequency-offset (whose magnitude is less than 30% of the symbol rate) and clock offset (whose magnitude is less than 50 parts per million(ppm)).

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Title	***:***	***Studies And Validation Of Solutions To The Forward Problem Of Fluorescence Diffuse Optical Tomography***
Author(s)	***:***	***Aggrawal Hari Om***
Roll No	***:***	***Y8104024***
Supervisor(s)	***:***	***Naik Naren& Sircar Pradip***

Abstract

Diffuse Optical Tomography (DOT) and its variants such as Fluorescence DOT (FDOT) are powerful methods for non-invasive early cancer detection. The solution of this nonlinear inverse problem typically requires an accurate solution of the optical propagation model for the measured excitance on the boundary given the spatial distribution of the medium's optical parameters (i.e., the solution of the forward problem). This necessitates a thorough validation of the solution schemes for the forward problem including comparison with experimentally measured quantities for a medium of interest. In the present work, a comparative study of forward model solutions using Monte Carlo, analytical and FEM based schemes has been done with the experimental excitance values obtained from phantoms for known optical parameters. We have also implemented the calculation of Jacobian matrices which is necessary for the inverse optimization schemes using the method of adjoints. FEM and Monte Carlo solutions have also been set up for fluorescence DOT problem. Numerical codes have been developed for the FEM solutions, while the standard Monte Carlo routine, MCML has been modified for fluorescence Monte Carlo procedures.

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Title **:** ***Resource Optimization for H.264 (SVC) Unicast /Multicast Wireless Video Communications***

Author(s) **:** ***Khanna Nitin***
Roll No **:** ***9104039***
Supervisor(s) **:** ***Jagannatham Aditya K***

Abstract

Future mobile applications using Fourth Generation (4G) wireless communication technologies such as LTE, WiMax and UMTS are based on the transmission of rich multimedia content. There is an ever growing demand for multimedia applications such as Standard / High Definition (HD) video streaming, on-line 3D gaming, multi party video conferencing, surveillance, etc. 4G wireless communication technologies in today's world are characterized by applications involving high quality and reliable delivery of multimedia content. Thus, wireless video communications is inseparable from the context of 4G wireless technologies. The H.264 Scalable Video Coding (SVC) standard enables layered compression, flexible extraction, transmission and decoding of partial bit streams to provide video services with varied temporal, spatial or fidelity resolutions to heterogeneous wireless end users. SVC provides functionalities such as graceful degradation in lossy transmission environments as well as bit rate, format, and power adaptation. Therefore, SVC has huge potential in multimedia communications which can be utilized in resource constrained arena of wireless communication systems. We studied various types of scalabilities of SVC standard by simulating a No. of example scenarios using Joint Scalable Video Model (JSVM), a reference software codec for H.264/AVC (SVC). In this context we consider a paradigm for sum video quality maximization for unicast and multicast video transmission to heterogeneous multimedia wireless clients. The media server is constrained by the processing overheads required for tasks such as video encoding, bit-stream extraction, packetization, etc. It is demonstrated that the above problem of video quality maximization is well represented by a constrained optimization framework. Further, this model can be readily extended to include Admission Control based Quality of Service (QoS) considerations in multicast transmission. Based on the system model, we present a closed form solution for frame rate allocation and a comprehensive algorithm for sum video quality maximization. As the solution obtained from the above mentioned technique contained non integer values for frame rates, it had limited applicability. To overcome this problem an optimal solution constrained to integer valued frame rates was derived by framing this scenario in form of dynamic integer programming problem (knapsack / capital budgeting scenario). It was demonstrated that the optimum solutions obtained using proposed integer frame rate allocation scheme were superior when compared to content agnostic equal frame rate allocation scheme. We further present a novel scheme for sum video quality maximization in the context of H.264 scalable video coding (SVC) based multicast video transmission to multimedia wireless clients with varied channel qualities. In a conventional multicast scenario, the multimedia server is constrained by the channel capacity of the worst-channel user in the multicast group. Our scheme

optimally partitions the multicast group into two sets for transmission of the base and enhancement scalable video layers. Thus, this avoids constraining the base station by the rate of the worst rate user, leading to a significant enhancement of net video quality. Our scheme is based on the Medium Grain Scalability (MGS) technique of H.264 (SVC) standard and we demonstrate that the optimal partitioning can be computed by adopting a Gradient Ascent (GA) based approach and solving a series of convex optimization problems. We compare the results obtained using the rate partitioning scalable video based scheme with a fair rate static video scheme in which the quantization step size is constrained by the worst channel user rate and demonstrate the superiority of the proposed scheme. A Real-time Transport Protocol (RTP) based streaming system was developed and implemented in a client-server model using Universal Datagram Protocol (UDP) of TCP-IP protocol suite. We used JRTP - an open source RTP library and Posix threads library in designing a multi-threaded real time video streaming system. We implemented an unbounded buffer using single link list and two worker thread model to delineate encoder/decoder from RTP socket handling. The shared data integrity was maintained using binary semaphores (mutex) in a producer-consumer model.

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Title	**:**	***Adaptive Power, Bit And Subcarrier Allocation For Ofdm Based Cognitive Radio With Primary User Queue-Awareness***
Author(s)	**:**	***Kumar Ayush***
Roll No	**:**	***Y9104010***
Supervisor(s)	**:**	***Banerjee Adrish***

Abstract

Cognitive radio refers to a form of wireless communication which bases its transmission and reception parameters on the cognition of its environment and offers efficient utilization of spectrum, reduced power consumption, and better sharing of resources among peers. Also, Orthogonal Frequency Division Multiplexing (OFDM) has been identified as suitable technique for implementation of cognitive radio due to its high flexibility with respect to the transmitted signal's spectral shape, thus making it possible to selectively leave a set of subcarriers idle. The users to which the frequency bands have been licensed are called primary users and the opportunistic users which tap into the spectral holes left by primary users are called secondary users. In this research work, we mathematically analyse the power, bit and subcarrier allocation problems for the three different paradigms of cognitive radio networks, namely interweave, overlay and underlay. We have analysed the effect of queuing of packets in primary user occupied subcarriers. The primary user subcarrier queue is not occupied all the time. This means that primary user transmission does not cause interference to the secondary user receiver all the time but only for a fraction of time which is given by utilization factor. This fact is incorporated in the performance analysis of various adaptive power, bit and subcarrier allocation schemes in this thesis. Through numerical simulations we show that the power allocation scheme proposed that incorporates the effect of primary user occupancy through the utilization factor of the queue results in higher secondary user throughput compared to the case when we do not use this knowledge. We also present some low-complexity but sub-optimal algorithms for realising the power, bit and subcarrier allocations and evaluate the performance of the corresponding allocation schemes through numerical simulations.

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Title : *Near Optimum Detection Of Tcm Signals In Coloured Noise*
Author(s) : *Singh Prem*
Roll No : *Y9104056*
Supervisor(s) : *Vasudevan Kasturi*

Abstract

Noise is one of the basic factor that sets limits on the communication system. For coded signalling, Viterbi Algorithm is optimum for signal detection in white noise. However in many practical situations noise is coloured or correlated. The sources of correlated noise can be the non-ideal detectors, pre-amplifiers and sampling circuits. The conventional detection schemes used in AWGN noise prove to be suboptimal techniques in the presence of such correlated noise. The optimum detection techniques for both uncoded and coded (Trellis coded) signals in additive coloured Gaussian noise (ACGN) have been derived earlier in the literature. For detection of coded signals in coloured noise, the linear equalizer-predictive Viterbi algorithm (LE-PVA) is the optimum scheme. The whitening property of the prediction filter is the basis for the optimal performance of the LE-PVA. However the performance degradation in the practical LE-PVA is mainly due to the fact that the LE trained using the LMS algorithm may not converge to the global minimum and perfect estimates of the autocorrelation of the error signal at the T-spaced sampler output are not available. In this thesis, we simulate a near ideal LE-PVA, which consists of a filter matched to the received pulse, a T-spaced sampler, a near optimum T-spaced equalizer followed by PVA, whose predictor coefficients are computed from theory. Note that it is not possible to implement the ideal LE-PVA since it requires an infinite length T-spaced equalizer and an infinite length predictor. We apply the proposed technique i.e. the near ideal LE-PVA, for two different channels and demonstrate its performance improvements over the practical LE-PVA.

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Title ***:*** ***Study of Different Pilot Patterns for Channel Estimation in OFDM Systems***

Author(s) ***:*** ***Ganesh Muthu***

Roll No ***:*** ***Y9104037***

Supervisor(s) ***:*** ***Vasudevan Kasturi***

Abstract

With the rapid growth of digital communication in recent years, the need for high speed data transmission is increased. Moreover, future wireless systems are expected to support a wide range of services which includes video, data and voice. OFDM is a promising candidate for achieving high data rates in mobile environment, due to its resistance to multipath fad-ing and ISI, which is a common problem found in high speed data communication.Wireless OFDM- is currently used and proposed for several broadcasting applications .The modula-tion in OFDM may be differential or coherent.Coherent modulation requires channel esti-mation which gives better performance but with relatively more complex receiver structure. Pilot Symbol Assisted Modulation is used to achieve reliable channel estimates by transmit-ting pilots along with data symbols. In this thesis, we will analyze different pilot patterns in terms of SER (Symbol Error Rate) and investigate a new scheme for transmitting pilot symbols in wireless OFDM systems. The reduction in the number of pilots will reduce the overhead in terms of transmission power and also improve bandwidth efficiency. It has been found through simulation that there is a definite improvement in the SER when a particular pattern of pilot symbol is adopted for wireless OFDM system.

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Title	***:***	***Simulation Study of Turbo Coded OFDM over frequency selective fading channels***
Author(s)	***:***	***Ray Choudhury Om Prakash***
Roll No	***:***	***Y9104053***
Supervisor(s)	***:***	***Vasudevan Kasturi***

Abstract

The main problems of reliable data communication in the wireless environment is the distorting multipath fading channel and Additive White Gaussian Noise (AWGN) noise. These impairments can distort the transmitted signal severely and thus leading to Inter Symbol Interference (ISI). So the reception becomes erroneous and the Bit Error Rate (BER) increases. Orthogonal Frequency Division Multiplexing (OFDM) or multicarrier communication is a recent technique used to mitigate ISI introduced by the distorting frequency selective fading channel. The earlier approaches used to combat ISI are based on Equalization and Maximum Likelihood Sequence Estimation (MLSE). Though MLSE is the optimum detector, its complexity grows exponentially with the channel length. Equalization has a low complexity but is suboptimal. OFDM essentially bridges the performance gap between MLSE and Equalization at a reasonable complexity. In this thesis, we attempt to study the performance of uncoded and turbo coded OFDM signal transmitted through frequency selective Rayleigh fading channels having uniform power delay pro le. The channel is assumed to be static for one OFDM symbol and varies randomly from one symbol to the next. Simulation results are presented for rate 1/3 and rate 1/2 turbo code

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Title : ***Optimal Diversity Power Allocation for Video Transmission in 4G OFDMA and MIMO Wireless Systems***

Author(s) : ***Mahajan Sohil***
Roll No : ***Y6927475***
Supervisor(s) : ***Jagannatham Aditya K***

Abstract

In our research we propose novel algorithms for optimal power allocation especially suited for video transmission in 4G OFDMA and MIMO wireless systems. In this context, we propose hierarchical video decomposition based optimal diversity subcarrier power allocation for video transmission in Orthogonal Frequency Division Multiple Access (OFDMA) wireless systems. The proposed video decomposition scheme is based on Discrete Wavelet Transform (DWT) and Hierarchical Block Motion Algorithm (HBMA) for spatial and temporal hierarchical-layering of the video. One of the key features of this schemes is that it employs partial channel state information (CSI) feedback based on the order statistics of the allotted subcarriers in the OFDMA system. This significantly decreases the overhead over reverse link and saves bandwidth. Thus it reduces complexity and enhances throughput. We demonstrate that the optimal power allocation can be obtained as the solution of an ordered subcarrier based overall received video distortion minimization problem and further illustrate that it can be formulated as a constrained convex cost minimization problem. We provide a closed form solution of the optimal power vector under suitable approximation and illustrate that cost minimization reduces to a polynomial root computation problem. In the context of Multi Input Multi Output (MIMO) wireless systems, we propose novel algorithms for singular mode diversity order based optimal power allocation speci cally for video transmission. We employ the paradigm of hierarchical video decomposition into spatial and temporal layers to exploit natural ordering in the diversity of the MIMO channel singular modes. We consider a practical MIMO system with the CSI not known at the transmitter. We employ a codebook based quantization at the receiver to feed back the index of the corresponding quantized beamforming vectors to the transmitter. Thus this scheme signi cantly reduces overhead over reverse link. We demonstrate that the proposed power allocation scheme can be formulated as a constrained optimization problem and the optimal power vector can be computed by solving an iterative sequence of convex optimization problems. We present a closed form expression of the solution to each iterative step. Simulations are performed employing several video sequences over OFDMA and MIMO wireless systems and results demonstrate signi cant performance improvement of the proposed schemes over sub-optimal equal power allocation schemes. Also, these performance enhancements are visually illustrated through picture quality comparison of the frames of decoded video sequences.

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Title	***:***	***Unsupervised Multimodal Methods for Speaker Diarization in Meeting Environments***
Author(s)	***:***	***Maheshwari Anurag***
Roll No	***:***	***Y6927097***
Supervisor(s)	***:***	***Hegde Rajesh Mahanand***

Abstract

Unimodal speaker diarization which uses information present in the speech signal alone has formed a vital part of speaker diarization research. On the other hand multimodal speaker diarization using both audio and visual modalities has also received wide attention in recent days. However when multiple microphones are used in the acquisition of distant speech as in meeting environments detection of number of sources also assumes importance along with speaker diarization. In this thesis a novel multi sensory and multi modal approach is proposed for detecting the number of sources and speaker diarization. We also analyze a speci_c multi sensory approach which uses a correlation matrix of the microphone array output to detect the number of sources. This analysis hitherto unused in the speaker diarization context is carried out to enable a fair comparison of the proposed multimodal method to unimodal methods in general. The proposed multimodal approach is unsupervised and uses only the signal subspace computed from the multi microphone inputs for a frame wise audio feature extraction. The visual features are computed from the frame wise motion features. Further a mutual information between the audio and visual features is computed. This mutual information is used to satisfy a condition in mutual information which can be used to and the active speaker. The knowledge of the active speaker in the current frame is used in the subsequent speaker diarization process. The technique is unsupervised, multi sensory and multimodal and therefore combines the utility of all the three techniques. The performance of the proposed unsupervised mutimodal method on speaker diarization is evaluated on two databases. The Clemson University Audio-Visual Experiments (CUAVE) database and the multimodal test bed data recorded at MiPS lab, IIT Kanpur are used in this context . The results obtained are encouraging as exhibited by the Detection Error Rate (DET) and the Receiver Operating Characteristic (ROC) curves, in comparison to the conventional unimodal approaches. The proposed unsupervised multimodal approach is further extended to applications like speaker detection and diarization in a distributed speech processing scenario to factor in bandwidth constraints. These experiments are conducted in contemporary teleconferencing environments where the bandwidth is of premium. The results of speaker detection and speaker diarization are reasonably better when compared to conventional distributed speech recognition results in similar environment.

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Title	***:***	***A Combined Approach to Speaker Verification using Claimant-Specific Phoneme Models***
Author(s)	***:***	***Arora Deepanshu***
Roll No	***:***	***Y6927157***
Supervisor(s)	***:***	***Hegde Rajesh Mahanand***

Abstract

A Speaker Verification system in general authenticates the identity of a claimant speaker by verifying the identity-related information in his or her spoken utterances. This information mainly relies on the voice characteristics of an individual and often times the performance of the SV system degrades due to high variability in the speaker's voice content. The other criteria that can be used in an SV system are the idiosyncrasies of the claimant speaker and the way in which one produces verbal content. Associated with these speech content variabilities is also the score variability which is considered a major reason for performance degradation in a speaker verification system. This thesis proposes a combined approach which uses both the speaker level and the phoneme level information to speaker verification. The approach draws on multi language speaker information captured from English and Hindi speakers using the universal background model (UBM) as the alternate hypothesis. More explicitly two new techniques based on phoneme level information have been proposed in this thesis. The first method is called the cohort set selection and normalization approach (PBCS-NORM) that improves upon the traditional cohort set selection approaches by using verbal information at the phoneme level. The second method is a late fusion method that uses majority voting after the normalized scores have been obtained. Both the proposed approaches are compared to the traditional normalization approaches like T-NORM, CWCS-NORM and the standard GMM-UBM based methods for both clean and noisy speech at various SNR. The performance evaluation of the proposed techniques is carried out on the TIMIT database and a Hindi database. The results are compared using the detection error tradeoff (DET) curves, the decision cost functions (DCF) and the equal error rate (EER) values. It is observed that the proposed approaches result in significant performance improvement over the aforementioned conventional methods.

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Title **:** ***Application of Compressed Sensing to Synthetic Aperture Radar Imaging***
Author(s) **:** ***Mohapatra Bibhuti Bhusan***
Roll No **:** ***Y9104011***
Supervisor(s) **:** ***Gupta Sumana***

Abstract

Synthetic Aperture Radar (SAR) provides high resolution imaging of a target or scene for various applications like target classification, automatic target recognition, ground mapping etc.As SAR operates in the Radio-Frequency(RF) domain,it overcomes the nighttime limitations of optical cameras, and the cloud-cover limitations of both optical and infrared imagers. The quality of a radar image is measured mostly by the resolution of the image. The range resolution depends on the transmitted bandwidth i.e. higher the bandwidth of the transmitted waveform,greater the range resolution achieved.But a high-bandwidth received waveform requires an Analog-to-Digital-Converter(ADC) with high sampling rate to perform digital pulse compression. The cross-range (azimuth) resolution depends on the size of synthetic aperture and the sample spacing in the azimuth domain.High azimuth resolution requires a closer sample spacing and thus a higher PRF for a given platform velocity .But a high PRF decreases the maximum unambiguous range of the Radar and also puts higher load on the transmitting device. So to create a high resolution map, the required sampling rate (based on the Nyquist criteria) is demanding in both, range and azimuth. Compressed Sensing(CS) is a recently introduced concept about acquiring a sparse or compressible signal in the most e cient way possible with the help of an incoherent projecting basis.CS theory asserts that one can recover sparse signals and images from far fewer samples or measurements than that used by conventional methods based on Nyquist sampling criteria. This thesis work presents a novel scheme for SAR image formation based on Compressed sensing to reduce the sampling requirements to a great extent in both range and azimuth. Using this approach we can transmit a fewer number of pulses and sample the received signal at a much lower rate than required by a conventional SAR and still can produce image of the same quality.The idea is to shift the load from sampling device to more powerful digital signal processor.This in turn results in signi cant reduction in data storage requirements and enables imaging over much wider swaths

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Title : *Design and Development of an Intelligent Meeting Capture System Using Speaker Localization and Visual Feedback*

Author(s) : *Tiwari Dheerendra*

Roll No : *Y6927160*

Supervisor(s) : *Hegde Rajesh Mahanand*

Abstract

A Meeting Capture System is an important tool for knowledge manage- ment. The degree of automation achieved in meeting capture systems till date, depends mainly on expensive hardware and provides limited selectiv- ity for data capture. There are two kinds of meeting capture systems. The rst type of systems is ring camera based wherein all the speakers are in a frame at a given time. The second type of systems is designed based on one dimensional direction of arrival information are prone to errors in direction of arrival estimation. However the advantage of the latter is its ability to selectively capture only the active speaker leading to e cient design of in- telligent and active meeting capture systems in real time teleconferencing environments. This thesis presents a complete design and development of a robust, real time intelligent meeting capture system based on two dimensional DOA es- timation and visual feedback. It improves the existing DOA based active meeting capture systems by using concepts from control theory. A visual feedback is used as a measure for error correction in speaker's azimuth and elevation angles as in done in conventional feedback systems in control the- ory. Other improvisations proposed in this thesis include a Generalized Cross Correlation based DOA estimation algorithm which is customized for a typical active meeting room scenario, wherein the speakers are meant to sit in a hemispherical three dimensional space with a video-conferencing unit setup in the other half. Visual feedback is generated on the basis of a map- ping between image-frame dimensions and real object dimensions. Based on this mapping the shift in speaker co-ordinates is traced back to error correction values in the two dimensional DOA estimates namely, azimuth and elevation angles. A novel geometrical method for range estimation is used for this purpose. A prototype real time intelligent meeting capture system has been designed and developed as part of this thesis. The performance evaluation of the intelligent meeting capture system is evaluated by conducting experiments on simulated real-time meetings con- ducted at the MIPS Lab multi modal test bed. The experimental results are presented as average error distributions (AED) for two dimensional DOA estimates (azimuth and elevation). The experimental results obtained for range estimates are also presented as AEDs. The experimental results on visual feedback are illustrated as error measures in face localization from the actual location of the face. The proposed system gives reasonable better performance compared to existing methods of meeting capture with a worst case response time of 3 seconds to locate the speaker exactly at the center of the image-frame in real time operation.

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Title	***:***	***Audio-Visual Information Fusion Methods for In-Car Speech Recognition***
Author(s)	***:***	***Vimal Pratik***
Roll No	***:***	***Y6927339***
Supervisor(s)	***:***	***Hegde Rajesh Mahanand***

Abstract

Modern day cars are equipped with functionalities that provide for seamless hands free in-car communication for people on the move. This includes communication via voice. Robust In-car speech recognition has therefore been in the realm of active contemporary speech recognition research. On the other hand, the performance of conventional speech recognition systems is known to degrade rapidly in noisy environments. Hence robust speech recognition systems have used various audio-visual methods to improve performance in noisy scenarios. The primary method of utilizing information in the audio-visual modalities is to fuse the information present in each modality. Several early and late fusion schemes have been used in this context. In this thesis a novel audio visual fusion method based on early fusion has been proposed and evaluated for robust in-car speech recognition. It also leverages on an existing late fusion method based on Dempster-Shafer theory. The proposed early fusion method fuses audio visual features at the frame level using visual feature interpolation. The process of interpolation is non trivial since the sampling rates of the visual and audio modalities is considerably dfferent. In this thesis the frame wise audio features are computed on a mel frequency scale. Video features are extracted from the lip region of the active speaker using non-linear scale histogram of the pixel values of each frame. The audio and visual features are then fused using a novel scheme of interpolation called cubic spline interpolation (CSPLINE-AV). The proposed CSPLINE-AV scheme is then used as front end for an in-car, audio visual speech recognition system. The decision level fusion mechanism used by this in-car speech recognition system is based on the Demspter-Shafer theory. The performance of the proposed system is evaluated on an In-Car database collected and transcribed at MiPS Lab IIT Kanpur. The database is developed on similar lines as the GRID corpus, which is an audio visual corpus developed by the university of Sheffield. Experimental results of audio visual speech recognition are presented for both the in-car speech database and the GRID corpus. The results of the proposed fusion methods are found to be reasonably better when compared to other conventional fusion methods for audio visual speech recognition.

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Title : ***Development Of Electronic Accessories For Atomic Clock***
Author(s) : ***Mishra Shitanshu***
Roll No : ***Y9104069***
Supervisor(s : ***SinghYatindra Nath&Pradhan Swarupananda***

Abstract

The objective of this project is to develop a system whose output frequency is highly stable. For this purpose we are using the atomic energy levels of rubidium atom as a reference for frequency stabilization of laser. First we have gone through various methods associated with locking a laser diode to Doppler broaden spectra of rubidium atom. Then we will look into atomic clock development based on Electromagnetically Induced Transparency (EIT). We have gone through various methods associated with this. Then we have developed a lock in amplifier and servo controller for locking the laser diode to Doppler broaden spectrum of rubidium atom.

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Title : *Multiview Video Coding using Layered Depth Image (LDI) and View Synthesis*

Author(s) : *Padma Santhosh Kumar*
Roll No : *Y9104064*
Supervisor(s) : *Gupta Sumana*

Abstract

Multi-view video consists of multiple videos of the same scene captured by several cameras at different locations. It comprises rich 3D information of the scene and has applications in new types of visual media such as free viewpoint video (FVV), free viewpoint TV (FTV) and 3DTV, as well as in surveillance, sports matches, games, and virtual reality (VR). Since the data size of multi-view video (MVV) increases linearly with the number of cameras used, it is necessary to compress MVV data for efficient storage and transmission. Hence the need for multi-view video coding (MVC) to realize the above applications. In this thesis work we use the concept of Layered Depth Image (LDI) to represent and process MVV with depth. It is an efficient image based rendering (IBR) technique generally used to represent three-dimensional objects with complex geometries. LDI contains several attributes together with multiple layers at each pixel location. Using LDI, we could reduce the large amount of MVV data to a manageable size by exploiting the spatial redundancies among multiple videos, and reconstructed the original multiple viewpoints successfully from the constructed LDI. View synthesis can also be thought of as a technique for achieving compression. It is a technique to generate intermediate virtual views from the MVV data. Instead of transmitting all the views as in simulcast, a subset of views can be transmitted and the remaining intermediate views can be synthesized on the receiving end. LDI approach has been used to code the subset of views transmitted. Generation of virtual images require the parameters of virtual camera that are calculated from the existing camera parameters. Using the virtual camera parameters and the MVV data, virtual views are created. But the main problem encountered in view synthesis is the occurrence of ghosting artifact. This artifact causes degradation of image quality and is visually unpleasant. In this thesis, we have proposed a new technique for removal of the ghosting artifacts. The experimental results validate the efficacy of the proposed method in successfully removing the ghosting artifacts

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Title	:	***Parametric Spectral Ratio Methods for Detection and Recognition of Whispered Speech***
Author(s)	:	***Mathur Arpit***
Roll No	:	***Y6927110***
Supervisor(s)	:	***Hegde Rajesh Mahanand***

Abstract

Conventional Automatic Speech Recognition systems use a large amounts of speech data which account for gender, accent and in many cases environment variations. However all such standard speech databases comprise of what is called the neutral mode of speech. The neutral mode of speech is different when compared to other natural modes of speech like emotional, shouted or whispered speech. This difference leads to a very poor recognition performance of the existing speech engines on natural modes of speech. Hence adaptation of existing speech engines to handle these natural speech modes in futuristic designs like automated hospital care and smart homes assumes importance in contemporary speech processing research. We propose a system that addresses the problem of handling different natural modes of speech by taking whispered speech as an example. The focus of the proposed system is on having an improved recognition system with minimal changes in the existing recognition engine. Towards this objective a new parametric spectral ratio method is proposed. This method computes the ratio spectrum of the Linear Prediction (LP) and the Minimum Variance Distortionless Response (MVDR) methods. The smoothed ratio spectrum is used to detect whispered segments of speech within neutral speech segments effectively. The proposed LP-MVDR ratio method exhibits robustness at different SNRs as indicated by the detailed experiments conducted on two databases. The proposed method also performs reasonably better than the conventional methods for whisper detection. In order to integrate the whisper detection method into a conventional speech recognition engine adaptation methods based on the MLLR are used to implement a mismatched train-test speech recognition system. The HMMs corresponding to neutral mode speech are adapted to the whispered mode speech data which are essentially detected from the proosed ratio method. The performance of this method is first evaluated on whispered speech data from the CHAINS database. The second set of experiments are conducted on the cell phone corpus of whispered speech collected at MiPS Lab IIT Kanpur using a set up that is very similar to the one that is used commercially for handling public transactions. The proposed whisper speech recognition system exhibits reasonably better performance as compared to the conventional systems compared in this work. These results indicate the possibility of a whispered speech recognition system for cell phone based transactions

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Title : *Throughput Maximization For Secondary User In Different Cognitive Radio Paradigm Employing Arq Protocol*

Author(s) : *Agrawal Gaurav*
Roll No : *Y9104017*
Supervisor(s) : *Banerjee Adrish*

Abstract

Spectrum is a public resource allocated by government to users for ex: TV channels, Cellular, unlicensed bands, radio. However it is found that some frequency band are mostly occupied, some are partially occupied and other are heavily used. This inefficient utilization of limited spectrum enforces the researcher to introduce the concept of cognitive radio in order to utilize the spectrum holes. The important feature of cognitive radios for wireless communication systems is the capability to optimize the relevant communication parameters given a dynamic wireless channel environment. Cognitive radio was proposed to promote the spectrum utilization by opportunistically exploiting the existence of spectrum "holes". However due to impairments on the radio fading channel, a packet can be received in error at the intended receiver hence require retransmission. Hence, ARQ protocols are employed in the system to provide error control. Meanwhile, cooperative relay technology is regarded widely as a key technology for increasing transmission diversity gain in various types of wireless networks, including cognitive radio networks. In particular, a source-destination link applying this protocol might hand over possible retransmissions to one of the available neighbour terminals that were able to decode the original transmission. Thus, the cooperative transmission is prescribed only if needed in an opportunistic fashion. This work addresses user coexistence in cognitive radio system while using an Automatic Repeat Request protocol that controls the transmission QOS in terms of throughput. In this research work, we mathematically analyze stable throughput for the three different paradigms of a cognitive wireless communication system, namely interweave, overlay and underlay under possibility of detection error & false alarm due to unpredictability of wireless channels. We analyze the cognitive link and primary link as interacting queue systems. We do stability analysis of an equivalent dominant system for the case of interacting queues with Loynes' stability criteria. Based on the studies, we derive optimal power allocation schemes for two links. This work also deals with protocol design for cognitive cooperative systems with single primary & secondary user for interweave cognitive radio. Analytical results show that the throughput of the whole system is greatly increased by exploiting the benefit of cooperative relay. The proposed protocols are studied from a queuing-theoretic point of view and the mathematical analysis is complemented by various performance evaluation results, which demonstrate the accuracy of the theoretical approach. In addition to that, our numerical results show that the Selective Repeat protocol results in higher secondary user throughput with or without feedback errors compared to the case when we use Go- Back-N & Stop and Wait

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Title	***:***	***Capacity Analysis Of Femto-Cell Based Cognitive Radio In A Two-Tier Network***
Author(s)	***:***	***Gupta Nitish Kumar***
Roll No	***:***	***Y9104051***
Supervisor(s)	***:***	***Banerjee Adrish***

Abstract

Recent research in wireless communication has shown that lot of spectrum bands remains underutilized most of the times. Cognitive radio has contributed in this regard for the opportunistic utilization of these underutilized bands by the unlicensed users. Cognitive radio (CR) was proposed in an endeavour to allow opportunistic use of unutilized licensed resources, also called spectrum holes, by sensing the available spectrum. However with the increasing cognitive users the capacity of cognitive users tends to degrade as the spectrum available per cognitive user decreases. Femto-cells are considered to be a promising solution for this growing demand of spectrum by indoor users as well as outdoor users. Also, Femto-cells can be incorporated with Cognitive Radio concept to utilize the spectrum resources e ciently. Femto cells deployed within a macro-cell region can be allocated split spectrum, partial shared or completely shared spectrum. In split spectrum as well as partial shared spectrum the capacity of the system is low but the interference introduced to either of femto-cell and macro-cell is minimum. So to improve the capacity of the femto-cell network co-channel deployment is considered with interference reduction techniques. In this thesis work, the problem of sub-carrier allocation, and power allocation is addressed in an OFDM-based two-tier femto-cell network comprising of a conventional macro-cell and multiple femto-cells .The macro-cell users nearby femto-cell causes signi cant interference to the femto-cell users as well gets interfered by the femto-cell transmissions. So, we propose an convex optimization method for solving sub-carrier and power allocation problem in under- lying femto-cell network. Lagrangian multiplier method is used in this regard to solve convex optimization problem for allocating sub-carriers to di erent users in a femto-cell and loading power on those sub-carriers in such a way that total interference to macro-cell users is under acceptable levels. Our objective is to maximize the multiple femto-cell users throughput by jointly allocating sub-carriers to users within a femto-cell, and also power allocation, under the constraints of cross-tier interference between macro-cell and multiple femto-cells as well as interference among femto-cells. Then, based on this sub-carrier and power allocation the performance of the two-tier network is analysed with capacity as the performance metric.

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Title	***:***	***Writer Dependent Cursive Handwriting Synthesis from Offline Samples***
Author(s)	***:***	***Shah Jaykumar Khushal***
Roll No	***:***	***Y9104068***
Supervisor(s)	***:***	***Venkatesh K S***

Abstract

This thesis presents a novel offline approach for training, modeling and synthesizing cursive handwriting in the subject's writing style. First of all, the subject's handwritten text document is scanned and converted into digital format, which forms the input to our system. Our system follows a sequential procedure for handwriting synthesis, which is carried out in two phases: (1) Training and (2) Synthesis. In the training phase, the digital document is passed through various preprocessing steps to obtain character skeletons representing the structure of character glyphs. The salient control points are extracted from an individual character skeleton followed by extraction of the feature points that represent key points in a vector graphics model. Our feature point extraction method employs recovery of temporal information from static handwritten character images while using an offline training procedure. In order that the synthesized text look realistic and more personal, a novel and effective approach for introducing natural variations in character glyphs is proposed in this thesis. The feature points of different samples of the same letter are matched followed by development of gaussian models parameterized according to the degree of deformation in the corresponding character glyphs. In the synthesis phase, character glyphs are generated from the feature points using cubic spline interpolation. A connection between each pair of adjacent characters is made, in case of cursive writing, using ligature strokes generated with the help of Bezier curves. Words and lines are composed aligning characters and words, respectively. Some randomness is added in alignment parameters to impart liveliness to the synthesized handwritten-like text. This complete information is then stored in standard PDF format, which can be read, printed and mailed from almost anywhere. Finally, the performance of our system is evaluated by subjective tests that demonstrate the capability of our system to produce style preserving cursive handwriting with pleasant visual quality.

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Title **:** ***Autocorrelation based spectrum sensing for OFDM over frequency selective fading channels***

Author(s) **:** ***RanjanSaurabh***
Roll No **:** ***Y9104066***
Supervisor(s) **:** ***Chaturvedi Ajit Kumar***

Abstract

Spectrum sensing is one of the most important requirements of cognitive radio (CR). It enables CR users to identify frequency spectrum bands currently available for transmission. In this thesis, we study the performance of autocorrelation based spectrum sensing technique for orthogonal frequency division multiplexing (OFDM) systems over frequency selective fading channels. Autocorrelation function exploits the cyclic prefix feature of an OFDM signal to detect the presence of a primary user. Generalized likelihood ratio test (GLRT) is performed on the autocorrelation function to obtain the sufficient test statistics. Analytical expressions for the threshold and the detection probability are derived and compared with the simulation results. It is found that autocorrelation based detection technique is computationally efficient and provides robustness against noise power uncertainty. Further, we apply this technique to hard and soft decision based cooperative spectrum sensing framework. The effect of bit error probability on the detection performance of a general hard decision fusion rule is studied. Subsequently, transmit beamforming based optimal combining (OC) and equal gain combining (EGC) soft decision fusion schemes are discussed under the constraint of total transmit power on all CRs active in the network. Finally, we compare hard decision and soft decision fusion based detection performance

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Title : *Deblurring and Depth Estimation From Defocused Video Sequences*
Author(s) : *Elubudi Rajeev Kumar*
Roll No : *Y9104058*
Supervisor(s) : *Gupta Sumana*

Abstract

In a typical imaging system, a camera is used for imaging the objects in a scene. It consists of the sub-systems: lens, diaphragm, shutter and recording media. All these subsystems introduce certain degradations to reduce the quality of the image and cause blurring. In addition, the blurring may also be caused by defocus, optical distortions, object motion or atmospheric turbulence. Blur images have less information than sharp images and lead to several difficulties in image analysis and scene interpretation. Among all these degradations, defocus blur and motion blur are the primary sources of blur in real time vision applications. In this thesis, we consider the defocus blur case for video sequences. In case of defocus blur the object point appears as a circular patch on the imaging plane of detector. The image of this point on image detector is known as the Point Spread Function(PSF). The radius of this blur circle is known as blur radius. This is an important parameter using which we can deblur the blurred frames and estimate the depth of the object. The depth information of a scene is a vital cue for the purpose of scene interpretation in machine vision system. The depth is defined as the distance between object and lens of the camera. In this thesis, we proposed an algorithm to estimate the defocus blur radius of all the frames in a defocused video. For this we exploited the relationship between adjacent video frames captured by an out-of-focus moving camera. We implemented the algorithm using two adjacent frames and multiple frames of the video sequence. We observed that the accuracy in the estimation of the blur radius increased with the use of multiple frames. By using the estimated blur radius and the given camera parameters, we estimated the depth of the object. Using the estimated blur radius we calculated the Point spread function of the defocus blur and restored the defocused video sequences using standard restoration filters.

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Title : ***Low Cost Hand Based and Eye Based HCI Systems Using Visible-IR Inputs***

Author(s) : ***Venugopal Lakkur Gautham***

Roll No : ***Y9104020***

Supervisor(s) : ***Venkatesh K S***

Abstract

We have developed low cost techniques which can be used in the development of human-computer interface (HCI) systems, especially for amputees and paralyzed persons. The system is based purely on eye actions such as blink and gaze control. Using this system, one gets the equivalent of simple mouse functionality, - point and click (both left and right click, including double clicks) albeit at a rather low resolution. For initialization, the subject just holds his/her head still for a few seconds: using involuntary blinks that would naturally occur during this time, the system both locates the user eye pair as well as forms online templates of the open and shut eyes of the specific user, valid for the rest of that session. The located eyes are tracked in real time using template matching and histogram back projection. Reasonable amounts of head motion, carried out at reasonable speeds, are automatically detected and compensated. Automatic re-initialization of the system occurs if the user goes out of frame or excessively rapid head movement occurs. The eye blinks of the user are detected in a robust manner by computing the Bhattacharyya distance between the histogram of the eye in the current capture and learned histograms of both open and closed eye templates; hence the sys- tem knows for each frame whether the eyes are open or closed, or in transition. When the eyes are detected as more than 70% open, the iris tracker module is triggered. In order to detect the iris, the eye image is subjected to image conditioning to eliminate the eyelashes and eyebrow. The iris is detected on the basis of colour and shape, and the Hough transform is used on the region to determine its centre and radius. Iris detection is continuously repeated whenever the eye is open, so that it is continuously tracked. Also, computer vision techniques employing simple probability approach is presented to convert an ordinary computer screen into a touch screen using a low cost webcam. The webcam is positioned to see the whole screen, image-screen calibration is used to map the points on the image plane and points on the screen. Viewing angle correction is used to compensate for intensity gradient of LCD screen. During run time, using the colour models of background and fingertip, the fingertip is located in the image and the image position is mapped to cursor position on the computer screen.

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Title	***:***	***High Accuracy Silhouette Based Reconstruction With Conventional Optics***
Author(s)	***:***	***Gadde Koteswara Rao***
Roll No	***:***	***Y9104029***
Supervisor(s)	***:***	***Venkatesh K S***

Abstract

Silhouette based techniques have been widely used for 3D object reconstruction. We have described a method of camera calibration using mesh-grid pattern for silhouette based method. Using this method, the real-world dimension measurements of the reconstructed visual hull can be done. This can be very useful for visual metrology. But, there could errors in the silhouette based reconstruction, because of the depth issue. A novel analytical algorithm has been proposed to solve the depth issue. To reduce the depth caused error in reconstruction and to achieve high dimensional accuracy a novel iterative algorithm has been proposed. The proposed algorithm significantly reduced the error caused by perspective effects. Thus this method makes the silhouette based reconstruction more robust in terms of dimensional accuracies of the reconstructed object. Off centered symmetrical object reconstruction has been done in both orthographic and perspective cases by modifying the basic silhouette based reconstruction. In this process offset of the object and the central axis of rotation have been successfully estimated.

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Title : *Low Cost Scale And Affine Invariant (Sai) Descriptors For Point Tracking In Color Videos*

Author(s) : *Venkatapathigari Venkata Satheesh Kumar*
Roll No : *Y9104065*
Supervisor(s) : *Venkatesh K S*

Abstract

This thesis deals with a novel alternative method for matching feature points in color video frames. Point matching has a vital role in many of the present day computer vision applications such as tracking, image indexing and retrieval etc. The problem in image point matching is to find features and descriptors that are invariant to the transformations that the image is likely to be subjected to. Besides, the descriptors should be of low dimension so that their extraction and matching is inexpensive for real time applications. We propose here, a simple 3 dimensional point descriptor that is invariant to scale and affine transformations, called the Scale and Affine Invariant (SAI) descriptor. From the three color channels, we obtain enough equations to solve for the affine parameters. This is next integrated with the optimum scale selection principle to achieve scale invariance in addition to affine transformation invariance. Finally, to improve false positives performance in frame to frame point matches, we also incorporate local histogram matching. The proposed method compares favorably or neutrally with SIFT with regard to accuracy, but is demonstrably lighter in computation

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Title **:** ***Camera Motion Tracking And Automatic Background Foreground Segmentation Of Moving Camera Data***

Author(s) **:** ***Sudheer K***
Roll No **:** ***Y9104026***
Supervisor(s) **:** ***Venkatesh K S***

Abstract

Automatic background foreground segmentation of moving camera is a key component of an automatic visual surveillance system. In this thesis, we propose a method for automatic generation of background and foreground for image sequences captured from a moving camera. Also, a technique for estimation of global shift vector is disussed. The global shift vector between consecutive frames is computed using Edge based Method and SIFT and a comparative study is done on the performance of both the methods. The contour of moving object is generated after motion compensation and pixel subtraction between two consecutive frames. Background learning is done in presence of moving targets by using a n pass outlier filtering algorithm. Foreground moving targets are extracted by background subtraction and thresholding. Experimental results obtained demonstrate the effectiveness and robustness of the proposed method.

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Title : ***Low Complexity Algorithms for Adaptive Bit and Subcarrier Allocation in Downlink OFDM Systems***

Author(s) : ***Agarwal Abhishek***
Roll No : ***Y6927015***
Supervisor(s) : ***Chaturvedi Ajit Kumar***

Abstract

In this thesis, we consider performance improvements in orthogonal frequency division multiplexing (OFDM) systems by taking advantage of adaptive modulation. For single user systems, by changing the modulation schemes in different subcarriers we allot bits to subcarriers with the aim of minimizing total transmit power, while satisfying target data rate and maximum bit error rate (BER) constraints. We consider optimal bit-allocation schemes as opposed to faster sub-optimal methods. The existing optimal bit-allocation algorithms have linear, in target data rate, computational complexities. Our proposed algorithm reduces the complexity to O(N logN), (N is the number of subcarriers) independent of the target data rate. Next, we look at the problem of subcarrier allocation for multiuser downlink, OFDM systems. Adaptive modulation is also assumed in this problem. The joint problem is NP-hard. We propose a low-complexity sub-optimal algorithm to minimize the required transmit power while satisfying users' Quality of Service (QoS) constraints. Comparing the performance of the existing algorithms with our proposed approach, in the simulation results, we find that our algorithm performs close to the optimal solution using low complexity methods. Extending the above approach to the problem of multicell subcarrier allocation in MIMO-OFDM systems, which is similar to the aforementioned multiuser problem, we propose a sub-optimal algorithm to co-operatively allot subcarriers to users with the aim of minimizing other cell interference. Our approach provides considerable reduction in the total transmit power without requiring complete co-ordination between base stations which consumes considerable fraction of the capacity of the back-haul link between them.

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Title : ***Error Correction in Wireless Sensor Network using Multisensor Prediction Model***

Author(s) : ***Jaiswal Aman***
Roll No : ***Y6927046***
Supervisor(s) : ***Jagannatham Aditya K***

Abstract

In this thesis, we propose a novel approach for error correction in a wireless sensor network (WSN). We begin with discussing the existing approaches by using a single sensor for error correction. We explain the previous setup start- ing from detecting the errors, to building of data models and nally rectifying errors thus creating the foundation for our research. In our setup we consider a case where multiple sensors are collecting data. The rst stage comprises of the incoming data modeling using Vector Linear AR equations. The output of the predictor thus gives us the choice of choosing the predicted value or the incoming observed value. Novel decision algorithms are employed for choosing the right value(observed or predicted). The predictor is designed using the auto-correlation of the incoming data and the cross-correlation among the sen- sors. Temporal auto-correlation exploits the temporal redundancy present in the data. Spatial cross-correlation is employed because sensors reporting sim- ilar quantities have correlated outputs though they are at spatially di erent positions. Simulation models are built using MATLAB for the complete setup, starting from generating the input data using AR equations which simulate realistic data models. Errors are added to these input models to simulate erroneus observed data values. We apply our multisensor error correction algorithm to detect and correct these errors. We compare our results with previous research which employs single sensor approach and use root mean-squared (RMS) error performance criteria of the algorithms to do the compartive analysis. Finally, a special case is considered where one or more sensor malfunctions among the sensors. We develop an algorithm to detect this malfunctioning and to detect the time instant at which this event occurs. We propose procedures to rectify the problem of failed sensors. This procedure is applied after detecting the point of sensor breakdown

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Title **:** ***Doubly Constrained Minimum Variance Beamforming Methods Using A Non-Reference Anchor Array***

Author(s) **:** ***Shukla Arpit***
Roll No **:** ***Y6927111***
Supervisor(s) **:** ***Hegde Rajesh Mahanand***

Abstract

Conventional Minimum Variance (MV) beamforming imposes a constraint on the spatial filter in the direction of the desired source signal. However in the MV beam forming method there is no constraint imposed on the design of the spatial filter with respect to the direction of the noise sources. The contributions of this thesis are two fold. Firstly a doubly constrained minimum variance beamforming method is proposed wherein an additional constraint is imposed in the direction of noise sources. The first constraint ensures unity gain in the direction of the desired source signal as is done in conventional MV beamformers. The additional constraint ensures zero gain in the direction of the noise sources which are assumed to be finite herein. A constrained optimization solution is provided to this problem under a specified cone of confusion. The proposed method is able to regenerate the desired source signal even in the presence of highly correlated noise sources in contrast to the conventional MV beamformers. The second part of the thesis proposes a non reference anchor array (NRA) framework wherein a noise source in the direction of the desired source can be effectively removed using the doubly constrained beamformer. In the NRA framework a primary array is used to capture the desired source signal while an auxiliary array is used to capture the signal from the noise source. The methodology for acquiring clean speech from these distant arrays is also discussed herein. The placement of auxiliary array plays a crucial role in the acquisition of clean speech. Hence a strategy to optimally determine the location of the auxiliary (NRA) array is also described. The strategy for optimal placement of the NRA array is also described more specifically in the context of a cell phone. The performance of the proposed methods is evaluated by computing average error distribution (AED) of the direction of arrival estimates. Additional experiments on distant speech recognition are also conducted and the performance is evaluated in terms of the word error rate (WER). The experiments were conducted on the S-TIMIT and MONC databases for the case of two and three noise sources. The performance of the conventional MV method using the non-reference anchor (NRA) array is reasonably better than that of the conventional MV method without NRA. When the doubly constrained MV beamforming method is used in the non reference anchor array framework the performance increases further in terms of the word error rate (WER). The performance of the doubly constrained MV beamforming within a non reference anchor array framework is also compared to correlation based (GCC, GCC-PHAT) and sub space based methods (MUSIC) and is found to perform reasonably better in terms of the AED and WER.

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Title* : *Video Compression Using Super Resolution Techniques

Author(s)* : *Singh Pushpendra Kumar
Roll No* : *Y9104057
Supervisor(s)* : *Gupta Sumana

Abstract

Image Super-Resolution (SR) has attracted substantial attention in the image processing community in recent years. SR is the process of obtaining higher resolution images from several lower resolution ones, i.e. a process of resolution enhancement. The quality improvement is caused by fractional-pixel displacements between images. SR technique allows one to overcome the limitations of the imaging system (resolving limit of the sensors) without the need for additional hardware. In this thesis, we approach this problem from the perspective of compressive sensing. Furthermore, we examine the very recent research area of SR for compression, which consists of intentional down sampling (during pre-processing) of video frames that are to be compressed and the application of SR techniques (during post-processing) on the compressed sequence. In this context, we propose a novel video decoding technique using SR image processing. The proposed method reduces the spatial resolution of all (not specific) pictures in the input video sequence and produce a low-resolution video sequence. It then encodes the low-resolution video sequence using standard H.264/MPEG-2 codec. On the decoding side the first frame of spatially reduced pictures is converted to its original resolution with the proposed SR method; the other low resolution frames are subsequently super resolved using either novel motion interpolation technique or using key frame based technique respectively. The proposed methods reduces the bit rate further compared to the compression achieved by H.264/MPEG-2 codec, as it encodes the video sequence in which all pictures are spatially reduced. We achieve 25 to 35 percent compression with the average 2 db loss in PSNR. At fixed bit rate we achieve 2 db improvement in PSNR.

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Title	***:***	***A Highly Modular Power Converter Architecture for Renewable Applications***
Author(s)	***:***	***Ray Olive***
Roll No	***:***	***Y9104052***
Supervisor(s)	***:***	***Mishra Santanu Kumar***

Abstract

With the demand of energy growing at an ever-increasing pace, new unconventional sources are gaining importance day-by-day. These sources are meant to complement the traditional grid-based system in developed countries. However, these sources have the potential to be the primary energy sources providing uninterrupted power supply to the end-user. These so called Distributed Generation Systems (DGS) have the potential to be the best option in areas lacking a well developed and reliable grid system. One of the problems that hinder the use of such sources are the different forms in which these sources generate their energy. This forces the use of different converter architectures to interface the source with the end-user. This thesis takes up the problem and develops a unique converter architecture which can be used for most of the energy sources available. The converter topology is bidirectional in nature, and it can be a potential option for DC to DC, DC to AC, and AC to DC power conversion. This makes it suitable for usage in a DGS where multiple sources are present. The topology cascades a bidirectional quadratic buck/boost converter with a conventional voltage source inverter (VSI). The power converter architecture is analyzed, designed, and validated in this thesis. A 40 V DC to 500 V AC (pk-pk) prototype is developed and validated to demonstrate the concepts. A case-study of the implementation of this converter for a DC to AC application is described in detail along with its digital feedback implementation. The experimental results show a reasonable correlation between the theory and experiments.

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Title **:** ***ANN Based State Estimation for Power Systems, including Bad Data Processing by Particle Swarm Optimization***

Author(s) **:** ***Kumar Amit***

Roll No **:** ***Y9104005***

Supervisor(s) **:** ***Chakrabarti Saikat***

Abstract

Nowadays, with the increase in electricity demand, the power transfer capability of the power system need to be improved. To improve the power transfer capability, new components such as Capacitors, FACTS devices, DC lines or even new AC transmission lines are being introduced in the power system. With the introduction of these new components, the size and connectivity of the power systems are becoming large and complex, and thereby requiring the improved monitoring and control. With the advent of Phasor Measurement Unit (PMU), in the Supervisory Control and Data Acquisition (SCADA) system, the accuracy of system visualization and state estimation has improved. However it needs more improved methods in place of conventional methods to sufficiently use the large amount of synchronized data at a very high rate from the PMUs. When both the PMU measurements with the SCADA measurements are used for state estimation, it is called Hybrid State Estimation. Ill ~conditioning of the gain matrix and computational time needed are some of the limitations of the conventional Weighted Least Squares (WLS) State Estimation (SE) method. In this thesis, Radial Basis Function Network (RBFN) is used for the purpose of hybrid state estimation and enhanced visualization for power system. RBFN is a powerful tool for modeling the complex systems by observing some critical input/ouput patterns. For hybrid SE, an RBFN is trained, by providing both PMU and SCADA measurement vectors as the inputs and estimated state vectors obtained by WLS method as the targets. % For the purpose of enhanced visualization another ANN is needed to be trained, to which only PMU measurements are provided as the input vector and corresponding estimated state as the target vector. In this way by using this ANN state vector of the power system can be estimated even when only PMU measurements are available. The effectiveness of the proposed method is tested on the IEEE-14, IEEE-30, IEEE-57 and IEEE-118 bus systems. It is observed that this method can be used effectively for enhanced visualization of the power system by using the PMU measurements.

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Title : *Estimation of Fault Location on Transmission Lines using Synchronous and Asynchronous Measurements*

Author(s) : *V C Shravan Kumar Reddy*
Roll No : *9104079*
Supervisor(s) : *Srivastava S C&Chakrabarti Saikat*

Abstract

Recent developments in Synchrophasor measurement technology has prompted utilities to deploy it in the power system networks for wide area monitoring and critical protection and control applications, such as fault location. The Wide Area Monitoring Systems (WAMS) utilize Phasor Measurement Units (PMUs), which compute voltage and current phasors at faster rate with time stamp synchronized to the Global Positioning System (GPS). This thesis provides an overview of recent developments in the area of fault Location. An efficient algorithm for hybrid fault location for two and three terminal transmission system using Supervisory Control and Data Acquisition (SCADA) measurements as well as PMU measurements has been proposed. A concept of synchronizing angle has been introduced for combing the SCADA and PMU measurements. Also an existing algorithm for fault location for two and multi terminal transmission systems using only the PMU measurements has been studied in detail and its performance has been demonstrated on the two, three, four and multi (six) transmission system. Simulation studies show that the performance of the proposed fault location algorithm using hybrid measurements is quite close to the existing method based on only the PMU measurements. The average error of the fault location is less than 1% with the proposed method and its performance is almost unaffected by the type of the fault and the fault impedance.

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Title : *Reconfiguration of Distribution Systems Considering Reliability and Power Loss*

Author(s) : *Bugga Amanulla*
Roll No : *Y9104012*
Supervisor(s) : *Singh Sri Niwas &Chakrabarti Saikat*

Abstract

Electrical power utilities have recently become more concerned towards maximizing the reliability of power supplied to customers and reducing the system power loss, specifically at distribution level. One of the effective operational measures to maximize the reliability and to reduce the distribution system losses is through network reconfiguration, both at the planning and operational stages. The optimal distribution network reconfiguration is formulated to minimize the unreliability and power loss, which is a multi-objective optimization problem. This thesis has applied binary particle swarm optimization (BPSO) to solve this multi-objective optimization problem by converting it into single aggregate objective function, with the idea of well-known weighted linear sum of the objectives. Probabilistic reliability models are used in order to evaluate the reliability at the load points. An algorithm for finding the minimal cut sets is used to find the minimal set of components appearing between the feeder and any particular load point. An algorithm is proposed to convert the improper branch data generated by BPSO in each reconfiguration, into proper sequence of branch data, which is used to evaluate the distribution system load flow. The system power losses are evaluated through distribution load flow. The proposed methodology is applied on a modified 33-bus radial distribution system and modified IEEE 123-bus distribution system. One can choose the desired optimal configuration based on the requirement of maximum reliability or minimum loss or trade-off between these two, by varying the weights assigned to each objective function

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Title : ***An Application Of Average And Marginal Participation Methods For Transmission Pricing***

Author(s) : ***Usmani Najmul Islam***
Roll No : ***Y9104044***
Supervisor(s) : ***Singh Sri Niwas&Singh Anoop***

Abstract

The power industry in India is undergoing through a change from a regulated to a competitive industry. Deregulation has unbundled the power sector into three separate business- generation, transmission and distribution. Earlier the generation, transmission and distribution were highly subsidized under the ownership of public utilities. The enactment of Electricity Act, 2003 has opened the door for the private investment in Indian electricity sector. Now, all the three sector of electricity sector are mix up in public and private sector. The provisions of open access have been instrumental in the development of competitive power markets in India. The transmission system is the backbone for the development of an efficient and competitive power market. The transmission pricing has become important since, the same transmission network has to be used by different public and private power utilities. Although, at present almost the entire transmission network in India is owned by Central Transmission Utilities (CTUs) or State Transmission Utilities (STUs), transmission charges can't be subsidized for public utilities as it will discourage private investment in power sector and hence, will hamper growth of Indian power sector. To support fair competition among power producers, one important aspect is to treat the transmission of electrical energy as a separate business, since electrical energy would become a product which could be bought or sold and transported from one place to another. So it is important to evaluate the actual cost of transmission facilities and the cost of providing the transmission services to the users for transferring power over a network. Moreover, the transmission services are to be treated as open to all the market participants interested in trading of electricity. The postage stamp has been used until now for transmission pricing, which is simple and works well for small network, where there are no inter regional power flows. In the last decade, Indian power sector has grown rapidly and is expected to grow more rapidly in the next decade. The- inter regional flows have increased and is expected to grow in next few years. Postage stamp charges the flat amount to each user in a region as if all the users are using system equally which is unfair to the users. In postage stamp method of transmission pricing, users using the system heavily get subsidized power at the cost of other users, using the system lightly. The Central Electricity Regulatory Commission (CERC) has proposed usage based method like average participation and marginal participation method, which charges the users based on the usage of the network. These methods have some theoretical backing which is supported by the electrical laws governing the flows in transmission network. In this thesis, the postage stamp, average participation and marginal methods are applied to 39 bus New England System and Orissa network and results are compared. The average participation and marginal participation methods support the idea that generators should also pay the transmission charges as they are also using the transmission network. The distribution impact on various users on applying these two methods has also been analyzed in this work.

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Title	**:**	***Future Image Frame Prediction Of An Image Sequence Using Artificial Neural Network With Selected Features***
Author(s)	**:**	***Soni Neeraj Kumar***
Roll No	**:**	***Y9104046***
Supervisor(s)	**:**	***Verma Nishchal Kumar***

Abstract

A comparative study of different feature selection techniques for future image generation of an image sequence using simple Artificial Neural Network(ANN)model is presented. The input to this network is a, unified hyper-dimensional colour and spatiotemporal feature space of the given image sequence, so that every pixel is uniquely defined. Separate ANN is formed for Red(R), Green(G) and Blue(B) color components of each pixel respectively and the output is then tuned for the R, G and B colour component for each of the corresponding pixel of the future image to be generated. Feature selection techniques like Interaction Information(II) and Bhattacharya distance are used to improve the performance of the model. Comparison with other feature selection techniques like Principal Component Analysis(PCA) and Mutual Information(MI) is also carried out for performance evaluation. Image quality measures like Mean Structural Similarity Index(MSSIM) and Canny based Image comparison Metric(CIM) are used to evaluate the quality of the generated future images. Three different image sequences, first is of a cyclone, second is from a fighter landing on ship and third is again of fighter landing on airstrip, are used to verify the model and last six images of the image sequences were successfully generated

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Title **:** ***Analog Maximum Power Point Tracking Controller For Solar Photovoltaic Application***

Author(s) **:** ***Nag Soumya Shubhra***

Roll No **:** ***Y9104073***

Supervisor(s) **:** ***Sensarma Partha Sarathi & Mishra Santanu Kumar***

Abstract

.Due to the increased energy demand and environmental concerns associated with it, the distributed generation systems(DGS) based on renewable energy source(RES)s are gaining importance day by day. Solar energy, mainly Solar photovoltaic(SPV) energy, is one of the most promising RES because of its availability, pollution free nature and lack of rotating parts. Power extracted from a PV panel depends on solar insolation, ambient temperature and terminal voltage of the panel. Thus the panel terminal voltage needs to be regulated to an operating point at which maximum power is extracted. This technique of extracting maximum power by adjusting its terminal voltage is called maximum power point tracking (MPPT). This thesis takes up the problem of developing an analog, low-cost MPPT controller. A new MPPT technique is developed based on the well-known Perturb and Observe algorithm. Costly multiplier and Sample & Hold circuits are avoided by employing multiplication-less measurement of power change and low-cost analog Sample & Hold circuit respectively. Simulation of the proposed system is done in MATLAB/Simulink and ORCAD PSpice. Real time implementation of the proposed system is done on a laboratory prototype. Experimental results obtained are in conformity with the simulation results and hence the effectiveness of the new MPPT controller is verified.

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Title : ***Bang-Bang Modulated FACTS Stabilizing Controllers Based on Online Identification of Critical Modes***

Author(s) : ***Goel Gaurav***

Roll No : ***Y9104021***

Supervisor(s) : ***Srivastava S C&Chakrabarti Saikat***

Abstract

The low frequency oscillations have been mainly observed in power systems because of the presence of high gain of exciters, operating the system under stressed conditions and the contingencies. Conventionally, the off-line approaches, based on the linearized model of the non-linear power system, are utilized to assess the small signal stability of the system. These model based approaches, in general, involve time extensive computations and are not suitable for real-time assessment of the system stability. With the recent developments of synchrophasor based Wide Area Monitoring Systems (WAMSs), which can provide time stamped and synchronized phasor data at a high refreshing rate, it is possible to have a measurement based real time estimation of the system stability including the identification of the small signal stability. Power system stabilizers (PSS) have been widely used, as supplementary controller to the generator exciters, for improving the power system damping. The Flexible AC transmission Systems (FACTS) controllers, such as Static Var Compensator (SVC), Thyristor Controlled Series Capacitor (TCSC) and Static Synchronous Series Capacitor (SSSC), have been primarily used either for providing reactive power support or controlling the line real power flow. Besides their primary task, FACTS controllers can also be used for improving power system damping, when provided with additional supplementary controllers. This thesis utilizes a control scheme to mitigate inter-area oscillations in power systems using bang-bang modulation of FACTS signals. The control scheme is used to quickly attenuate the system's most dominant mode, identified using wide area measurements and Prony method. A method to determine controller's switching time has been suggested. A control strategy, utilizing line reactive power flow as the input signal to the SVC damping controller, has also been proposed. Moreover, the proposed control scheme allows several controllers to act simultaneously, to provide better damping. The proposed control strategy of stabilizing controller has been tested with SVC, TCSC and SSSC in MATLAB/SIMULINK. Simulation results on various test systems show that following the disturbance, the power system critical mode of oscillations are damped out effectively and the controller's performance is robust to changes in the system operating condition. Amongst the three FACTS controllers, better damping is achieved with the SSSC

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Title : *A Stereo-Vision Approach to 3D object grasping using the Barrett hand*

Author(s) : *Choudhury Chandrajit*
Roll No : *Y9104013*
Supervisor(s) : *Behera Laxmidhar*

Abstract

One of the basic problems in robotics is grasping. Given any object, deciding the grasping point on that object is a major challenge in robotic grasping. Extensive amount of work has been done in grasping of planar and symmetrical objects. There are some proposed methods for grasping of novel objects, using machine learning and statistical methods. In many works, grasping has been done using laser scanners. This research work is mainly focussed in vision based grasping. In this research work few existing methods in vision based grasping are implemented and analysed, and a new stereo-vision based grasping method is proposed. The proposed grasping algorithm is based on 3D surface reconstruction of the object. For 3D surface reconstruction of the object stereo-vision is used. In this context, the problem of stereo-correspondence is also discussed. A stereo-vision system has been set up, calibrated and homographic transformation is carried on the stereo-image pair. The major types of stereo-correspondence algorithms are discussed, few of them are implemented on the transformed image pairs. A method for stereo matching in a non-diffused lighting condition, with less computational cost has been proposed. Realtime experiments for grasping of symmetrical objects, like a cube and a ball, are carried on a robotic setup consisting of a stereo-camera pair, a 7 DOF robot manipulator and a Barrett hand. The methods and results of these experiments are discussed.

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Title **:** ***Multi-Criteria PMU Placement For Power System Observability***

Author(s) **:** ***Meena Nand Kishor***

Roll No **:** ***Y6927278***

Supervisor(s) **:** ***Chakrabarti Saikat***

Abstract

In wide area monitoring, protection and control (WAMPAC), Phasor Measurement Units (PMUs) are playing an important role. A PMU can measure the voltage of the bus and current for all the incident braches to the bus at which PMU is placed. Due to economic and technical constraints, it is not yet feasible to place the PMU at every bus in a power system. This thesis proposes a methodology for optimal placement of PMUs using binary particle swarm optimization (BPSO) in normal operating condition of the system. The proposed method is also used to place the optimal number of PMUs in the presence of convention measurements in the power system. There may be more than one PMU placement solution (corresponding to different locations for the same number of PMUs) for complete observability of the system. Proposed method considers two criteria to choose the best solution among these multiple solutions. One is to choose the solution, which results in the maximum measurement redundancy at the buses; and the second is to choose the solution, which results in minimum condition number of the gain matrix of the hybrid state estimator in order to improve the convergence properties of state estimator. The proposed method is successfully applied on various IEEE test systems. Power system network is spread widely. Generally it contains many geographically and operationally defined areas. Each area has its own state estimator to estimate the local states of the area, and these local states are transmitted to the central state estimator for central state estimation. The main advantage of multi-area PMU placement is that, central state estimator is able to get a reference angle for that area and it helps to improve the state estimation process. Chapter 4 in this thesis proposes a PMU placement methodology for multi-area observability. The proposed methodology applied on various IEEE test systems.

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