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Microelectronics, VLSI & Display Technology

Signal processing, Communication & Networks

RF Engineering & photonics

Power &Control

Title : ***Analog Circuit Fault Diagnosis by Selective Decomposition Using DC Fault Dictionary Method***

Author(s) : ***Parthasarathy S***
Roll No : ***null***
Supervisor(s) : ***Sharan R***

Abstract

Electronic equipment used in Army, have generally been developed on a hierarchical or modular structure. Repair and maintenance of these equipment also follow a well defined hierarchy. Though number of fault diagnosis techniques are in use in army, there is a need for having a fault diagnosis technique that could be used at any hierai'chy of maintenance. Keeping this requirement in mind, "Analog Circuit Fault diagnosis by Selective Decomposition using DC Fault Dictionary method" has been developed. This method would give a working solution which could be easily implemented for repair and maintenance of electronic equipment at different hierarchies with little modifications. The concept of Field Replaceable Unit(FRU) has been used to narrow down a fault to the lowest replaceable sub unit of a system. Further, this approach can be used both for linear and nonlinear circuits. Due to decomposition and identification of faulty subnetwork prior to application of fault dictionary method, faults have been adequately isolated. The on-line computation time will also be greatly reduced in comparison to a standard fault dictionary method as the data that would be processed has also been greatly reduced(i.e. the data pertaining to the faulty subnetwork is only processed). The effectiveness of this method in fault identification has been studied using two examples and the result has been found to be encouraging.

For more details click here **Back**

Title	***:***	***An Improved Subthreshold Region Model For Short Channel Mosfets***
Author(s)	***:***	***Thakare Shivraj Bhagwan***
Roll No	***:***	***9610460***
Supervisor(s)	***:***	***Dutta Aloke***

Abstract

As analog and mixed - signal VLSI designs are becoming popular, an accurate modeling of the subthreshold region for short channel MOSFETs is becoming extremely important. This region of operation is exploited in modern application areas, such as space electr onics, laptops, medical applications, communications, multimedia, etc. operation of the devices in their subthreshold region reduces the total power dissipation, which is crucial as currently the packing density is increased manyfold due to the rapid advan cements in the photolithography techniques. Experimental data on short channel MOSFETs reported in literature show that the subthreshold slope is a function of the applied drain voltage, which previous authors have failed to model. In this work, we have ta ken up an existing subthreshold slope model (which is independent of the drain voltage) and have modified it to include the effect of the drain voltage. Additionally, our model accounts for the effects of the effective channel length and the body voltage o n the subthreshold slope. We have also attempted to represent the fudge factor, used widely in the expression for the characteristic length, by a more physical representation than done before. The subthreshold slope model has been put in an existing drain current model, valid for all regions of operation, in order to obtain the DC characteristics. The simulated results are compared with those reported experimentally for MOSFETs having submicron channel lengths, with the minimum being as small as 0.075 µm. th e results show an excellent show an excellent match between the two

For more details click here **Back**

Title **:** ***Analog To Digital Conversion - A Neural Network Approach***
Author(s) **:** ***Thakur Manish***
Roll No **:** ***9610425***
Supervisor(s) **:** ***Sharan R***

Abstract

This work presents a neural network based analog to digital converter. A/D conversion is viewed as an optimization problem and hence solved by associating a cost function with the problem statement. A general class of recurrent neural networks have their e nergy functions matched with the cost functions defined above. These energy functions are associated with circuit elements which are just comprised of feedback resistors and amplifiers. Thus, from the energy functions we finally get the detailed circuit sc hematic needed to minimize the cost function we started with. A simple 4 - bit A/D converter was implemented to demonstrate the idea. Conventional architectures had the problem of incorrect conversion because of the network getting trapped in local minima. A modified circuit was proposed and implemented which ensured global attractivity to elude local minima

For more details click here **Back**

Title	**:**	***Implementation Of GUI And Non-Blocking Queueing Network Analysis For QNAT***
Author(s)	**:**	***Bhaskar D M***
Roll No	**:**	***9610404***
Supervisor(s)	**:**	***Bose Sanjay Kumar& Manjunath D***

Abstract

In This thesis we have implemented some well known techniques for solvingqueueingnetworks hat have only non-blocking nodes in the nr=networks in a user friendlysoftware package called Queuing Network Analysis Toll (QNAT). We have alsodeveloped some techniques to reduce fork-join queues that may or may not havesynchronizing queues and have made them available in QNAT.QNAT can solve openand closed queuing networks with single class jobs open, closed and mixed network withmultiple class job's QNAT does not allow jobs to change classes. Fore the analysis of the open networks we use the technique developed for networksof GI/G/m queues by Whitt. For closed networks we use well-known Mean ValueAnalysis (MVA) technique. For mixed networks that have open and closed classes ofcustomers, we first find the utilsation of the servers for the customers of the open classes.This utilisition is use to reduce the service rate (inflate the service demand) for the closedclasses of customers. We now consider only the closed classes of customer but with thereduced rate of service at the queues and solve the work using the MVA technique. Thesegive us the queues and then solve the network using the MVA technique. This gives usthe performance of the open classes of customers is then obtained from those of theclosed classes of customers In QNAT, fork-join queues can also be specified in the network. The fork-join queuemay or may not have synchronizing queues after the service queue in the siblings. Thenetwork may be open or closed. In all these case a flow equivalent service center (FESC)for the fork-join queue is obtained using suitable techniques. The fork-join queue isreplaced by this FESC and the result ant network is solved using standard techniques. Afork-join queue with synchronizing queue is not supported by QNAT for open networks.QNAT can also handle queuing networks that have finite capacity queues and theirimplementations are deleted in a companion thesis.

.

For more details click here **Back**

Title	***:***	***Distributed Bandwidth Allocation And Call Control For VP Based ATM Networks With Multi-Class Traffic***
Author(s)	***:***	***Gupta Sonia***
Roll No	***:***	***9610454***
Supervisor(s)	***:***	***Bose Sanjay Kumar***

Abstract

Broadband network based on ATM are expected to be the target technologies for futurecommunication systems. These networks will typically be able to provide multiplexingand flexible switching to support a diverse mix of call traffic between various O-D pairs.This thesis presents schemes for bandwidth allocation and call control that may beapplied to such networks under dynamically changing traffic load patterns. A flexiblerule-file based simulation approach has been proposed for studying such systems underdifferent algorithms by writing appropriate "rule file". These rule file are written asexternal ASCII text file using a language and syntax provided by us and the actualoperation of handling VP-based AT network with dynamically changing load conditions.The proposed algorithm utilizes information kept locally with the link of the network.This is used to efficiently assign adequate bandwidth to the various origin-destinationpairs depending on their requirements. This is also indirectly used for the call acceptancecontrol algorithm and to decide the routing of an incoming call. This distributedalgorithm has been studied through simulations and its performance was found to be verysatisfactory. Sample simulation results of this system have also been presented anddiscussed

.

For more details click here **Back**

Title **:** ***Design And Synthesis Of Asynchronous Digital Circuits***

Author(s) **:** ***Garg Ajay Kumar***

Roll No **:** ***9610401***

Supervisor(s) **:** ***Roy Subir Kumar***

Abstract

Asynchronous design has been an active area of research since early 60's. There has been a resurgence of interest in the design of asychronous circuits in recent years, due to their ability to eliminate clock skew problems, achieve average case perform ance, automatic adapt automatically to processing and environmental variations and provide a better technology migration route as compared to synchronous circuits. There are several implementation approaches for asynchronous digital circuits based upo n the delay model used. These models are the bounded delay model, delay insensitive model, speed independent model and the quasi delay insensitive model. In the present thesis, we describe the methodology for the designing asynchronous digital systems base d on a delay insensitive model using the Non Return to Zero (NRZ) event driven signaling protocol. We use this approach to design an asynchronous arbiter and the bus based interconnect structure for bundled data transfer. Desai [23] has given the different implementations of the bus based interconnect structures. But the implementation presented here has the advantage of reducing the bus width and the decoder complexity. Several methodology exists for synthesizing delay insensitive digital circuits e.g., trace theory and transforming asynchronous circuits as communicating processes (CP) to delay insensitive circuit using CSP language. We synthesize delay insensitive circuits from its behavioral description written in a hardware description language (HDL) such as Verilog and VHDL. The design is represented in a HDL and its control and data flow graph (CDFG) is generated. We describe two algorithms Event List Scheduling (ELS) and the Modified Event List Scheduling (MELS) for scheduling. Allocation and binding of resources in an asynchronous digital systems. We test these algorithms on several high level synthesis benchmark examples. Finally, we synthesize the various asynchronous circuit modules for a Xilinx FPGA target architecture

For more details click here **Back**

Title **:** ***Design And Implementation Of A Viterbi Decoder For A CDMA Modem***

Author(s) **:** ***Pandita Bupesh***
Roll No **:** ***9610408***
Supervisor(s) **:** ***Roy Subir Kumar***

Abstract

A major contribution of this thesis has been in the design and implementation of a Veterbi decoder. A Viterbi decoder is a very important block in any CDMA modem. The aim was to design a 19.2 kbps, 256 state Viterbi decoder with added capability of catering to higher input data rates. To the best of our knowledge none of the existing literature discusses Viterbi decoding implementation based on high level synthesis targeted for field programmable gate arrays (FPGAs). This has been the focus in the pre sent thesis. Besides the above, some of the issues such as organization of a path memory, decision memory, the decision memory reading techniques, and the clocking mechanism have been discussed. We have also retained the bit - synchronization information even though it made the normalization of the path metrics essential. We have used very fast subtracters to implement normalization of the path metrics. In this thesis we explore an implementation methodology for rapid prototyping using FPGAs. In des igning the Viterbi decoder we have used some of the features discussed in the literature; however, we have attempted to compare their merits with other existing techniques and also the advantages that have accrued because of their use. It has not been our attempt to give just another implementation of the Viterbi algorithm, but to create design guidelines which may be used in any future implementation of the decoder

.

For more details click here **Back**

Title **:** ***Multiparty Authorisation Protocol : Design And Performance Issues***

Author(s) **:** ***Chandravadan Nanavaty Niraj***

Roll No **:** ***9610428***

Supervisor(s) **:** ***Manjunath D***

Abstract

A transaction processing (TP) system is essentially a query based system and its fundamental executable unit is called transaction. An authentication mechanism is needed to identify the requester and the server. An authorisation protocol is required to con trol the access to the system resources. Many a time, a transaction may require autorisation from multiple parties. Such authorisations are called multiparty authorisations. In this thesis we investigate the process of multiparty authorisations, the design of a protocol for multiparty authorisation, various design tradeoffs and performance issues. The difficulties in mapping existing largely accepted broadcast protocols such as reliable broadcast protocols by Chang and Maxemchuck are highlighted. A fail sa fe multiparty autorisation protocol (MAP) is designed correct and deadlock free operation of the protocol is ensured using timeout the retransmission strategy. Certain design issues that can help reduce the number of messages transmitted and tradeoffs with the corresponding increase in delay that might result in are also investigated. Detailed investigations for simplified assumptions of identically behaving authorisation servers are presented

For more details click here **Back**

Title : *Gossiping In Multihop Radio Networks*

Author(s) : *Gupta Neeraj*
Roll No : *9610437*
Supervisor(s) : *Manjunath D*

Abstract

Gossiping is a specific information dissemination requirement communication networks.In gossiping each node in the network has information that it needs to communicate toeveryone else in the network. This problem has many practical applications incommunication networks where it is often necessary for each node to maintain up-to-dateinformation regarding processor loads, lengths of queues, routing tables etc., of othernodes in the network. Gossip algorithms have been studied for a variety of communication models like mailmodel, telephone model etc. However very little work has been done in understandingor developing gossip algorithms for radio networks. In this thesis we first study gossiping in communication networks by conducting anextensive literature survey of the various communication networks by conducting anextensive literature survey of the various communication models and the gossipalgorithms therein. We then describe the problem of gossiping in radio networks using agraph theoretic framework and propose some good algorithms for gossiping in both therestricted and general topologies. In the single frequency multihop radio networks that we consider, we assume that timeis slotted and the system is synchronous radio networks that we consider, we assume thattime is slotted and the system is synchronous. Moreover, in each time slot a station ispermitted to transmit only one message out of the several ones that it may have at thetime is allowed to transmit only one message out of the several ones that it may have atthe time it is allowed to transmit in the network. For general topologies three algorithmsfor gossiping have been proposed each having its own advantage s and disadvantages.We believe that third algorithm that we purpose, the Gather-Scatter algorithm, couldreally be among the best algorithms that can achieve gossip in radio networks. Thespecific topologies that we consider are the complete graph, star, ring and bus network sand these are chosen from the point of view of their applications in personal and indoorcommunication systems. We supplement the theoretical study of all our proposedalgorithms with extensive experimental results. Our experimental results are based on anexperimental model that is a realistic representation of real-life radio networks.

For more details click here **Back**

Title : ***Queueing Analysis Of Scheduling Policies In Copy Networks Of Space Based Multicast ATM Switches***

Author(s) : ***Sikdar Biplab***
Roll No : ***9610406***
Supervisor(s) : ***Manjunath D***

Abstract

In an ATM based BISDN network, multicast capability of the switches, in addition to their normal unicast operation, would be an inherent feature. The general structure of a space - based, space division multicast switch is that of a copy network followed by a routing network. The copy network generally uses a multistage interconnection network where the switch elements have broadcasting capability. Our interest in this thesis is on the queueing analysis and modelling of space based copy networks. The nature o f the copy network and the scheduling policy which determines the order in which the input ports are served play a major part in defining the overall performance characteristics of a multicast switch. In this thesis we first propose an exact model to calcu late the overflow probabilities in Lee's copy network, where the input ports are unbuffered. Next, we introduce buffers at these inputs and propose queueing models for the copy network for three scheduling policies - cyclic service of the input ports with an d without splitting of copy requests and acyclic service without fanout splitting. These models evaluate the copy network performance in terms of the average delay experienced by the copy requests. We also introduce a new parameter, the sustainable through put of the copy network, which is defined as the maximum load that can be applied to all the input ports without resulting in an unstable queue at any of the inputs. We also propose methods to calculate the sustainable throughputs. Our queueing models are verified against simulation models

.

For more details click here **Back**

Title : ***Analysis Of Various Types Of Blocking In Finite Capacity Queueing Networks***

Author(s) : ***Tahilramani Hema***
Roll No : ***9610415***
Supervisor(s) : ***Bose Sanjay Kumar& Manjunath D***

Abstract

Queueing Networks Analysis Tool (QNAT) is a versatile, user - friendly software package developed y us at the Indian Institute of Technology, Kanpur for the analysis of a large variety of blocking as well as non - blocking queueing networks. QNAT can han dle general configurations of open and closed networks of both finite and infinite capacity queues. This thesis deals with the analysis of the blocking queueing networks in QNAT. The typical blocking mechanisms that have been incorporated by us in QNAT ar e Transfer Blocking, Repetitive Service Blocking (Fixed or Random Destinations) and Rejection Blocking. Product - form Approximations have been used for the analysis of closed networks with transfer blocking. Maximum Entropy methods have been used for the an alysis of both open and closed networks with repetitive service blocking (RS - RD & RS - FD). A method based on markov chain techniques has been used to analyze open networks under rejection blocking. This thesis also presents a new approximation technique for the analysis of open networks with transfer blocking. This technique is based on reconfiguring the network with the addition of hypothetical nodes and modifying the network flows appropriately. A product - form decomposition of the modified network with Max imum Entropy based analysis for the individual queues is then carried out. The results of this method have been verified by comparing against those obtained from a Discrete - Event Simulator

For more details click here **Back**

Title **:** ***Parametric Estimation Of Non-Stationary Signals***

Author(s) **:** ***Sharma Rakesh Kumar***
Roll No **:** ***9610440***
Supervisor(s) **:** ***Sircar Pradip***

Abstract

In this thesis a new model is suggested for modelling non-stationary signals. The signal is modelled as an amplitude and frequency modulated signal. Autoregressive model based on modified covariance algorithm is used for estimation of autoregressive coefficients. These coefficients are used to find zeroes and thereby estimation of carrier and modulating frequencies. Estimation of modulation index is done using properties of Bessel function. Removal of strong individual subsignals, which interface with the estimation of other parameters is also discussed. The model is fitted on a noise corrupted computer synthesised amplitude and frequency modulated signal. The suitability of model for a real signal is also carried out by fitting a real electroencephalogram signal. The study demonstrates the suitability of the model, and elaborates the approach for estimation of parameters

.

For more details click here **Back**

Title **:** ***Parametric Modelling Of Speech Signal***

Author(s) **:** ***Saini Major Rakesh Kumar***

Roll No **:** ***9610439***

Supervisor(s) **:** ***SircarPradip***

Abstract

Complex AM and FM signal models can be used for representation of non - statinary signals such as speech [1,2]. Complex AM signal model has been found to be suitable for voiced speech phonemes [1], while complex FM signal model can be used for representation of unvoiced speech phonemes [2]. This study demonstrates techniques for faster estimation and computation of parameters of these models which can ultimately lead to on - line processing of data and ease in automation.

For more details click here **Back**

Title	***:***	***Instantaneous Frequency Estimation Of Speech Resonances***
Author(s)	***:***	***Rao N V Chalapathi***
Roll No	***:***	***9610427***
Supervisor(s)	***:***	***Rao Preeti***

Abstract

In the production of voiced speech the glottal source and the vocal tract interact, giving rise to variations in the formant frequencies and bandwidths over the duration of a single pitch period. Spectral correlation technique cannot be applied to ana lyze these variations, as they are limited by the available number of data samples. In an attempt to overcome these limitations, a new approach is tried in this work to track these variations of formant frequencies using an instantaneous frequency estimati on technique based on the analytic signal method. The estimate is unreliable near the glottal closure and opening instants while there is a clear increase in the frequencies of the first and second formants in the glottal open phase of some cases.

For more details click here **Back**

Title : ***Circuit Modelling Of Power Bipolar Junction Transistor***
Author(s) : ***Vij Vipul***
Roll No : ***9610464***
Supervisor(s) : ***Patil M B***

Abstract

A power bipolar junction transistor (PBJT) model is developed giving both static and dynamic behavior. The ordinary differential equations are derived based on the semiconductor physics. These equations take into account the recombination, charging and discharging process in the lightly doped collector drift region. The model is implemented along with associated circuitry, like base drive, load (resistive and inductive), by writing a C - program. The static behavior giving quasi - saturation characteristics can be simulated with this model. The transient behavior of the model is studied with different base drives, time steps and error margins. Also charge profile in the drift region is presented at different instants during turn on/off. Experimental results, wherever available, are plotted along with simulated results. The matching between the two validated the model developed. Also qualitative agreement with published work exists for various simulated results presented in this thesis work

For more details click here **Back**

Title **:** ***Techmig: A Layout Tool For Technology Migration***

Author(s) **:** ***Kar Pradip Kumar***
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Supervisor(s) **:** ***Manjunath D***

Abstract

With numerous foundry services being made available for implementing systems on VLSI chips and with the rapid strides that are being made in the fabrication technology of these chips. It is imperative, from the point of view of economics, to be able to migrate any design from one foundry specific technology to another, as well as, from a present generation fabrication technology to the next generation fabrication technology, within the same foundry. As the ability to integrate and pack more devices with in a die of silicon increases with every new generation of the fabrication technology, the complexity of digital systems realizable in a single chip has also grown by leaps and bounds. To circumvent implementation bottlenecks. A lot of research work has be en carried out to fully automate the design of VLSI chips under prespecified area and performance criteria. The focus of the present thesis is on an important of VLSI chip design known as physical layout design. The basic issue addressed here is the following: "Is it possible to port and reuse existing cells in a particular generation of a fabrication technology to a new, but evolving generation by using the mask layout descriptions of the existing cells?". As the creation and validation of the mask l ayout description of cells in any fabrication technology is a time consuming, error prone, tedious and costly process, it is important to be able to make the best possible use of the existing layout resources accumulated from the initial fabrication proces s. This process is known as technology migration. In this thesis, study of some existing software tools and algorithms that have been employed for technology migration has been carried out. Some new approaches, and the corresponding algorithms for ca rrying out technology migration has been proposed. These algorithms have been incorporated in a technology migratory, called TECHMIG. TECHMIG has been developed as a part of the thesis, TECHMIG takes as its input the mask layout description of an existing cell, expressed in the industry standard Caltech Intermediate Format (CIF), along with a set of design rules specific to a new fabrication technology or a new foundry, and a set of user specific design constraints. It then retargets the initial cell by pr oducing its mask layout description in CIF which is appropriate for the targeted fabrication technology or the foundry.

For more details click here **Back**

Title **:** ***Implementation Of Public Key Elliptic Curve Cryptosystems Over GF (2n)***

Author(s) **:** ***Mathur Manish***

Roll No **:** ***9610424***

Supervisor(s) **:** ***Siddiqui M U& Umesh Srinivasan***

Abstract

In some applications of public -key cryptography it is desirable and perhaps even necessary, that key size be as small as possible. Moreover, the cryptosystem just needs to be secure enough so that breaking it is not computationally feasible. Most of th e known public - key cryptosystems are totally insecure if the key size is restricted to about 100 - 150 bits. Recently, Lenstra demonstrated the feasibility to factorize a 450 - bit composite integer and La Macchia and odlyzko computed logarithms in the field o ver 192 - bit prime, while Gordon and mccurely were able to compute logarithms in F 2 401 . These results justify the unsuitability of RSA and EIGcmal schemes for the applications requiring smaller key sizes. A suitable candidate for such applications that rema ins is an elliptic cryptosystem that provide equivalent security as RSA and other systems but with a much smaller key length. The purpose of this thesis is to provide a practical implementation of these systems. With the increasing demand of smart card based applications, efficient software implementation of elliptic curve cryptosystems poses a challenge for the cryptographers and software professionals. In this thesis, we have made an attempt to implement them on Pentium and as well as on TMS320C4 0 digital signal processor using optimizing C cross compiler. The algorithm we adopted is the elliptic curve based ELGamal scheme over galois field GF(2 n). To obtain minimal complexity in computations, we used optimal normal bases for field arithmetic

For more details click here **Back**

Title	**:**	***Input Speech Identification Using Discrete And Semi-Continuouu Hmm***
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Supervisor(s)	**:**	***Ray G C***

Abstract

The task of a speech recognition system is to recognize the words spoken by all speakers in all environments. The first step of a speech recognition system is front end processing and the second is acoustic pattern matching. The job of a front end processo r is the convert the speech waveform to some type of parametric representation for further analysis and processing. The front end processor used in this thesis work is based on Linear Predictive Coding (LPC) analysis. Linear predictive coding analysis tech nique highlights a reliable and tractable representation of the speech signal spectra. With a given speech signal representation, acoustic pattern matching will detect and classify possible acoustic patterns, which can be phonemes, syllables, words or sent ences from speech signals. Acoustic pattern matching forms the central issue in speech recognition research. Most important progress has been achieved using technique based on dynamic time warping (DTW) algorithm, hidden Markov models (HMM), and neural net works. Hidden Markov modeling is a technique for the study of observed items arranged in a discrete - time series. The items in the series can be individually or continuously disturbed. The HMM has been shown to represent one of the most powerful statistica l tools available for modeling speech signals, and has been successfully used in automatic speech recognition systems. In this thesis work, two different methods for modeling of an isolated word in a HMM based speech recognizer have been used and compared . The first technique is discrete modeling, where the acoustic space is partitioned into some finite segments using vector quantization. The representation of the vector quantization codeword in the sample space is the centroid of the corresponding cell. I n the Discrete HMM, vector quantization (VQ) produces the closest codeword from the codebook for each acoustic observation. This mapping from continuous acoustic space to quantized discrete space may cause serious quantization errors for subsequent hidden Markov modeling. This quantization error can be avoided if the observations does not come from a finite set, but from some set of continuous points. This is called fully continuous HMM. But this method needs extensive training data and computation time. To overcome these limitations of discrete and fully continuous HMM, the VQ codebook can be modeled as a family of finite mixture probability density functions such that the distributions are overlapped, rather than partitioned. Each codeword of the codebook can then be represented by one of the probability density functions and may be used together with others to model the acoustic event. The use of a parametric family of finite mixture densities can then be closely combined with the HMM methodology. This met hod of modeling is called semi -continuous HMM. Training and testing of these two modeling methods have been carried out on 13 words recorded in noise free

environment. These words are first sampled at a frequency of 10kHz, and then they are quantized by using a 12 bit A to D converter. Then the end point detection algorithm is used for proper location of speech signal. Then by using LPC analysis sequence of observation vectors are obtained. Later, these observation vectors are used for training and testi ng of two modeling methods. One model per word is trained by using 5 utterances of the same word. During testing, the calculations of model likelihoods of all possible models are carried out: followed by selection of a word whose likelidhood is high est. It has been proved that the recognizer based on semi - continuous hidden Markov model is more robust than the one which is based on discrete HMM. Also recognition accuracy of the semi - continuous HMM is greater than the discrete HMM. But as far as comp utational time is concerned, the recognizer based on semi - continuous HMM needs extensive computation time as compared to discrete HMM. While simulating both these methods, it is observed that the most important part of the HMM based speech recognizer is t he estimation of model parameters. It is a straight forward approach for discrete HMM compared to semi - continuous HMM. Also is semi -continuous HMM estimation of Gaussian densities is computationally extensive and computation goes on increasing with the inc rease in number of probability densities. The initial values of the mean vector and covariance matrix for Gaussian densities have to be generated randomly. Depending on the testing results these values have to be modified. Also with the changes in Gaussian distributions, the model parameters also get changed. So far proper re - estimation of both, the Gaussian densities and the model parameters, a lot of computation is involved. The recognition accuracy obtained for discrete HMM is 98.46% while that for se mi -continuous HMM is 80.00%. This contradicts theoretical results. In theory, it is proposed that the recognition accuracy of semi continuous HMM is more than discrete HMM. This paradox is due to improper re - estimation of the model parameters of semi – cont inuous HMM and insufficient training data. Although, five utterances of each word have been used for training the HMM, these are not sufficient for proper re - estimation of model parameters. With large training data, model parameters can be estimated accura tely. Since, data recording and its digitization is a lengthy process, we have used only five utterances of each word.

For more details click here **Back**

Title : *Estimation Of Motion From Motion-Smeared Images*

Author(s) : *Thomas Vinu*
Roll No : *9610463*
Supervisor(s) : *Gupta Sumana*

Abstract

Motion estimation is an important topic in contemporary research in visual processing. The present day technology places constraints on camera shutter speed and the illumination level for adequate signal to noise ratio. Thus smear due to motion may occur in captured images. The motion - smear which is usually considered as a degradation in the image contains information about motion parameters and can be used for their estimation. In this thesis, a computational model is developed that estimate motion from m otion -smear in the frequency domain. The proposed algorithm is verified using computer generated images, texture images and a real time image. The proposed method is found to be cost effective and computationally simple. It can be used to estimate motion parameters especially in defence related areas where trade off exists between available illumination level and sensor shutter such as aerial reconnaissance under poor visibility conditions, underwater imaging etc.

For more details click here **Back**

Title : *Rate Based Pacing And Class Based Queueing For A Hybrid Gateway Server*

Author(s) : *Sharma Vivek*
Roll No : *9610467*
Supervisor(s) : *Manjunath D*

Abstract

Most of the Internet traffic is asymmetric in nature. This is the motivation for the development of systems that provide for a high bandwidth channel for direction in which there is a larger volume of traffic and a less expensive low bandwidth channel in the direction in which there is a low volume. A hybrid gateway server may be used to provide this kind of a service. In our design of this service, we split the TCP connection between the client and the application server at this HGS. Thus the HGS handles a large volume of traffic on the high bandwidth channel and manages the connections from many groups of users and is also expected to carry a wide variety of traffic. In this situation, the TCP throughput needs to be maximized and also shared fairly among the many users and types and traffic. In this thesis we address two problems that potentially lower the TCP throughput: odd/short - final - segment problem is handled by disabling Nagle's algorithm and slow - start - restart problem is addressed by implementing a rate based pacing of the TCP packets. The short - initial - segment problem is discussed but no solution is proposed. The high bandwidth channel is expected to carry many kinds of traffic each requiring different kinds of service from the network. We pr ovide a rudimentary quality of service to the traffic going out of the satellite Gateway server by implementing a class based queueing scheme in the device interface. Traffic through the SGS are divided into multiple classes, separate queues maintained for each class and bandwidth guarantee is provided for each class. The transmission scheduling algorithm also guarantees bounds on the time between consecutive services to a queue class. The implementations are tested and the results are presented

.

For more details click here **Back**

Title **:** ***Analysis Of Multicast Flow Control***

Author(s) **:** ***Singh Arun Kumar***
Roll No **:** ***9610403***
Supervisor(s) **:** ***Manjunath D***

Abstract

A growing number of computer network applications require the use of a reliable multicast service to disseminate data from sender to a set of receivers, typically called a multicast group. Delivery of data packets from the sender to the receivers takes place through a multicast delivery system. An efficient implementation of routing of packets in such a delivery systems is by the use of a multicasting distribution tree. Further, a flow control and reliable delivery mechanism is necessary between the send er and the receivers. A window flow control mechanism is an obvious choice for the flow control mechanism. In a multicasting framework, a separate window flow control mechanism would be to have separate connections between the sender and the receiver s and operate individual windows between the sender and each receiver. This is because in window flow control, acknowledgments are needed from all the receivers to ascertain the correct receipt of packets at the destinations. However, if the group is large , this leads to an acknowledgment implosion at the sender. To avoid this problem most multicast protocols use a hierarchical arrangement of the receivers that for tree rooted at the sender. The non leaf nodes will act as designated receivers for the subtre e rooted at these nodes and will reliability and flow control for the nodes of this sub tree. In this arrangement separate windows can operate independently between successive layers of the tree. In this thesis we present queueing models for analyzing the delays in such multilayered system. Since exact analysis is difficult, approximations are suggested and the results from these approximations are compared against simulation results

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For more details click here **Back**

Title	***:***	***Remote Tutor : Organising, Recording And Playback Of Lectures***
Author(s)	***:***	***Kumar N V Sudheer***
Roll No	***:***	***9610455***
Supervisor(s)	***:***	***Manjunath D***

Abstract

Remote Tutor is an Internet teleseminaring tools on MS Windows platform. With thisapplication, a tutor can lecture and interact with students through voice, slides graphicsand text. The application uses whiteboard, a window shared among the group, to displayPostScript slides, exchange graphics and multifont text in a multicast environment.An earlier teleseminaring tool, SlideTalk was implemented on the Win 3.11 platform.However, the voice delivery on this was defective. In this thesis, SlideTalk is upgradedto Remote Tutor with many new features and the voice component has been redesignedyielding telephone quality voice, as specified for the application.An important feature of Remote Tutor is the capability for offline lecture organization,recording and storage. The tutor can organize a lecture in units organization, recordingand storage. The tutor can organize a lecture in units and store these units in a predefinedsequence, before lecture delivery using a GUI based organizing aid developed as a part ofthis thesis. We also allow recording and storage of a lecture and delivery of a storedlecture. Also, the tutor could view the organization of the lecture during delivery.The application is designed completely under Object Windows environment and uses IPmulticast for communication between members. UDP is the underlying transportprotocol

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For more details click here **Back**

Title : ***Lightweight Communication Protocol For Distributed Computing***

Author(s) : ***Pandey Ritvik Ranjanam***

Roll No : ***9610447***

Supervisor(s) : ***Manjunath D***

Abstract

Applications like distributed computing need frequent and intensive transaction of data over a communication network. Schemes like Message Passing Interface (MPI) provide communication libraries, in addition to others, to effect distributed computing over a network. Most implementations of these libraries use TCP/IP protocols for transport and network layer functions while the libraries themselves reside in the application layer. Since TCP/IP is designed to work reliably in very large networks too, it is bound to be slow and inefficient for small, high performance, reliable networks. Due to this, the transport layer becomes the bottleneck in the computing speed a considerable amount of time is spent in communicating. This thesis designs and implements a lightweight communication protocol, LeghtCommunicator, to substitute the heavyweight TCP/IP stack in distributed computations over small, high speed LANs. The substitute offers the same reliability characteristics as that of TCP/IP but has a processing delay of half that of TCP/IP. In the design of LightCommunicator, we assume that all communication is over the same LAN.

For more details click here **Back**

Title : ***Estimation Of Local Bandwidth For Non-Stationary Signals***

Author(s) : ***Goyal Sumeet***
Roll No : ***9610456***
Supervisor(s) : ***Sircar Pradip***

Abstract

Local bandwidth has been estimated for the non - stationary signals. For estimating the local bandwidth the real time data (signal) has been approximated using orthogonal polynomial series. Assumption has been made that in the bounded region the highest freq uency will dominate in the higher order derivatives (say higher than sixth order derivative). Derivatives have been estimated using orthogonal polynomial approximation. Before testing the efficacy of this method on real time signals, it has been tested on sine -waves and mixture of sine - waves of different sets of frequencies. This method of estimation of local bandwidth has been applied to real time non - stationary signals of ECG and EEG. The results have also been cross - verified by auto - regressive method of frequency estimation

For more details click here **Back**

Title : *Design Of A Low Bit-Rate Video Coder: An Object-Based Approach*

Author(s) : *ReddyNarsi V*
Roll No : *9610461*
Supervisor(s) : *Gupta Sumana*

Abstract

In this thesis we have implemented the object based analysis-synthesis coder (OBASC) for the encoding of images at very low data rates. A coder based on this concept divides an image sequence into moving objects. An object is defined by its uniform motion and described by motion, shape and color parameters. The parameter sets of each object are obtained by image analysis. Using the parameter sets an image is synthesized by model-based image synthesis. The important step in image analysis is the estimation of 3D motion parameters of the object. In order to accomplish this we have proposed a new method which has given better results than the gradient based method. In comparison to block-based hybrid coder, object based approach requires the additional transmission of shape parameters. The transmission of shape information avoids the mosquito and blocking artifacts of a block-based coder. Compared to block based coder, the quality of the reconstructed images was increased for the same alloted bit rate. The coding of parameters is not attempted in this thesis. Assuming that the color parameters are coded at a bit rate of 1.0 bit/pixel, it can be concluded that the overall bit rate required for head and shoulder image sequence for a frame rate of 10 Hz is well below 64 Kbit/sec.

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For more details click here **Back**

Title : ***An Implementation Of The Mpeg Audio Coding Layer 3 Algorithm***

Author(s) : ***Patwardhan Pushkar P***
Roll No : ***9610436***
Supervisor(s) : ***Rao Preeti& Umesh Srinivasan***

Abstract

MPEG -1 Audio Layer 3 is a perceptual audio coding algorithm capable of providing near - CD -quality at data rates as low as 56 kb/s per channel. In this work, a software implementation of the MPEG - 1 Audio Layer 3 encoder has been carried out for single channe l audio signals sampled at 44.1 kHz. Based on suggestions in the informative section of the MPEG - 1 ISO Standard Draft Document, detailed algorithms have been developed to implement the key functional blocks of the layer 3 encoder viz., the filterbank, psyc hoacoustic model and dynamic bit - allocation scheme. Algorithmic procedures to incorporate the special features of the layer3 algorithm including dynamic windowing, variable bit - assignment based on perceptual entropy estimates and lossless coding of quantis ed data have been developed. Investigations which lend an insight into this complex algorithm, including the interaction of various functional blocks, have been carried out using suitable test audio signals. These investigations also serve to generate a be tter understanding of the critical task of tuning the various perceptual and bit - allocation parameters to achieve the required audio quality at any specified bit - rate. Informal subjective listening tests have been carried out at bit - rates from 56 kb/s to 9 6 kb/s using a variety of audio material. While the decoded audio quality is indistinguishable from the original at 96 kb/s, a slight perceptual degradation is observed at 56 kb/s in certain cases.

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For more details click here **Back**

Title : *On The Analysis Of Optical Fibers With Arbitrary Refractive Index Profile*

Author(s) : *Jethi Narasingha*

Roll No : *9610430*

Supervisor(s) : *Sircar Pradip*

Abstract

Two fast and accurate numerical algorithms based on the orthogonal polynomial approximation and Chebyshev polynomial interpolation for obtaining the actual field solutions of a fiber with arbitrary refractive index profile are described. These solutions in turn are used to evaluate the group delay - time and fractional power flow in the core of the fiber. The numerical methods are compared in terms of their performances, namely, complexity and accuracy. The method based on the orthogonal polynomial approximat ion is very simple and convenient to implement on a computer and it has some other advantage compared to the Chebyshev interpolation method

For more details click here **Back**

Title : ***Study And Design Of Mpeg-2 Transport Stream Multiplxer Of Private Data Broadcast***

Author(s) : ***Murty D V S N***

Roll No : ***9610410***

Supervisor(s) : ***Srivathsan K R***

Abstract

Digital TV and audio broadcasts are poised to replace the current analog TV broadcast for entertainment, news and other one way broadcast applications. Broadcast channels provide high quality with good SNR at affordable costs. Digital TV provides two new opportunities besides current broadcast services – first, emerging services based on video - on - demand and second, data broadcast related applications. Data broadcasting from a central server site with low bandwidth interactive channels to clients provides n ew broadcast applications in the form of hybrid network to Internet and electronic commerce to homes at affordable cost. In this thesis review of MPEG - 2 standard for video, audio and private issues compression and multiplexing is first carried out. Th en a detailed study on the issues involved in the imbedding of private data over the MPEG - 2 transport stream multiplexing is presented. Finally, some measurements of the proposed private data multiplexing in the transport steam are presented based on trial implementation over a software MPEG - 2 transport stream multiplexer

For more details click here **Back**

Title : *Studies On GPS And Its Applications In Networking*
Author(s) : *Hemachandra R*
Roll No : *9610443*
Supervisor(s) : *Srivathsan K R*

Abstract

This thesis is a study on some of the ways in which GPS can be put to use in networking. In particular we have employed GPS as an accurate source of UTC time ad used this precise GPS timescale to synchronize the clocks of different systems in a computer network. Accurately synchronized clocks are an essential ingredient for the smooth functioning and reliable trouble-shooting of a distributed computer network. The NTP has been widely used to achieve a reliable network time synchronization through a hierarchy of time servers. We have implemented a stratum one NTP Primary time server on a Linux platform using GPS receiver. This time server can provide an accurate time reference to all the systems within the institute network, or outside, if necessary. The thesis also carries a short review of the structure of GPS signals, the commercial grade GPS receiver outputs and their variance in standalone and differential modes

For more details click here **Back**

Title : ***Delayed Feedback Arrival Rate Control (With Hysteresis) In Single Server Finite Buffer Queues***

Author(s) : ***Verma Sonal***
Roll No : ***9610453***
Supervisor(s) : ***Manjunath D***

Abstract

A generic feedback congestion control scheme in a packet switched communication network would use arrival rate control. Congestion is typically indicated by the buffer occupancy level in the packet queue at the node and this information is fed back to the source. The sources on receiving the information regarding the level of congestion in system adapt their sending rates according to some predetermined algorithm. Our interest in this thesis is in constructing queuing models for arrival rate control in a jingler server, finite buffer queue in which there is a non zero delay between the instant at which congestion sets in and the time at which sources react to the information by way of modifying their arrival rates. We allow two rates to be used in the s ystem and provide for hysteresis in the rate control scheme. It is assumed that there are two classes of packets and during congestion, one of the classes is disabled . A n important contribution of this thesis is the analytical modelling of deterministic delay in the feedback path. A n embedded Markov chain with the markov points embedded at the packet arrival instants when the switchover is not in prog ress and at the switchover beginning and completion instants is defined and solved. W e also consider the case of exponentially distributed dela y in the feedback and obtain a c ontinuous time Markov chain model for this system. W e derive the expressions for various performance parameters like total throughput, throughput of each class, blocking probability of each class, average queue lengths, average delays and switchover rates for both deterministic and exponential feedback delay for different values of the buffer si s ze, switchover delays and arrival rates.

For more details click here **Back**

Title : *Development Of Control Laws For An AFCS Of A Helicopter*

Author(s) : *Thampi Geetha K*
Roll No : *9610412*
Supervisor(s) : *Sule Virendra Ramakant*

Abstract

For more details click here **Back**

Title **:** ***Order Reduction And Vibration Minimisation In Helicopter Fuselage***

Author(s) **:** ***MathewsAnita***
Roll No **:** ***9610402***
Supervisor(s) **:** ***Sule Virendra Ramakant& Venkatesan C***

Abstract

The i eduction of helicopter vibration remains a major challenge to the rotorcraft designer The present study addresses the problem of foimulation and design of a multi-mput-multi output closed loop vibiation control scheme, of a 3 D flexible fuselage model based on the concept of ACSR(Active Control Of Structural Response) The sensor locations for vibration measurement are chosen from a optimal set of locations measuring high levels of vibration Initially the dynamic model of the full order system is decomposed into flexible and rigid body mode subsystems A reduced order model for the flexible mode system is formulated using balanced realisation based-order reduction technique Contiollei is then designed for this reduced model using factorisation theory and internal model principle for distuibance rejection The simulation study showed good vibration reduction in the fuselage and gearbox Increasing the number of sensors seemed to reduce the magnitude of control forces required for vibration minimisation The choice of low vibraioiy level node location for sensor placement gave unsatisfactory results It is obseived that sensor locations have a significant influence on the level of vibration 1 eduction in fuselage structure r

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For more details click here **Back**

Title : **Comparison Of Front End Features In Hmm Based Digit Recognition**
Author(s) : **Sinha Rohit**
Roll No : **9710443**
Supervisor(s) : **Umesh Srinivasan**

Abstract

In this work, we have evaluated the performance of continuous digit recognizer on two most popularly used front - end features, namely LPC - and MEL - cepstral features so as to study the robustness of the above said features for digit recognition task. For this purpose we have implemented a HMM based continuous digit recognizer using development environment provided by CSLU -Toolkit. Our study finds that MEL - cepstral feature provides slightly better work and sentence level accuracies compared to LPC - cepstral features. Our study also indicates that for a fixed product of number of states and number of mixtures per state in a model, the models with higher number of state result in better work as well as sentence level accuracy. In the later part of this work, we have reported a study on the relationship between any two speakers, which is important from the point of view of gaining an insight into the development of speaker - independent speech recognition systems

For more details click here **Back**

Title : ***Characterization Of Open-End And Gap-Discontinuties In Single And Coupled Planar***
Author(s) : ***Jhally H S***
Roll No : ***9610414***
Supervisor(s) : ***Biswas Animesh& Shukla Rajiv Kumar***

Abstract

Full wave analysis has been done for characterisation of open-end in single and coupled lines, gap discountinuity and off-set layered gap, in suspended substrate as well as MICs environment. Based on this full wave analysis, the software has been developed to model the open-end in single and coupled lines, gap-discountinuty in suspended substrate and MICs environment.

For more details click here **Back**

Title : **Study Of Cylindrical Dielectric Resonator In Tmo1$ Mode And It's Applications To Microwave Filters**

Author(s) : **Dhingra Sandeep**
Roll No : **9610448**
Supervisor(s) : **Biswas Animesh& Shukla Rajiv Kumar**

Abstract

In this thesis, a comprehensive study on dielectric resonator operating in TM o1 δ mode, kept in various types of environments, has been done. The analysis presented here, is applicable to isolated DR, DR post, DR in a cavity, DR in cylindrical metal wave guide bilow cut - off suspended substrate etc. various parameters, required for the design of microwave circuits including DR for millimeteric wave applications, have been studied and analytical formulations have been developed for them. The resonance freque ncy of a cylindrical DR for TM o1 δ mode, is calculated using dielectric waveguide model (DWM), and some modifications over it. The fields, energy distribution in all the regions of the structure are obtained. Q - factor has been found using the basic definiti on of Q, based on losses and energy relations. Inter - resonator coupling between two identical DRs for edge coupled and broad - side coupled cases have been found, using electric dipole model having capacitive coupling. Results obtained in this thesis, ar e in good agreement with the published results, where -ever available

For more details click here **Back**

Title	***:***	***Full Wave Analysis Of Microstrip-Slot Coupled Line On Isotropic And Anisotropic Dielectric Substrates***
Author(s)	***:***	***Palanichamy S***
Roll No	***:***	***9610450***
Supervisor(s)	***:***	***Biswas Animesh***

Abstract

Microstrip -slot coupled line is a type of transmission line which is widely used as a transmission medium in microwave and millimetric wave region. This structure has many advantages over other planar transmission lines. The propagation parameters like , normalized guide wavelength (0llg), phase constant (β), effective dielectric constant (∈eff) and characteristics impedance (Zo) are essential in the design of Microwave integrated circuits (MIC's) e.g. directional couplers, filters, ph ase shifters etc. thus the propagation parameters of microstrip - slop coupled line on isotropic and uniaxial anisotropic dielectric substrates are analyzed using full wave modal analysis method. The computational method presented in the thesis is general in nature and can be applied to unilateral fin lines and suspended substrate microstrip lines by changing the width of strip or slot. The numerical results for the frequency dependant propagation parameters have been obtained and presented. The use of micros trip -slot line for phase shifter application at microwave frequency has been presented

For more details click here **Back**

Title	**:**	***Broad Band Suspended Substrate Radial Line Stub - Analysis And Measurements***
Author(s)	**:**	***Joshi Ravi Kumar***
Roll No	**:**	***9610445***
Supervisor(s)	**:**	***Biswas Animesh &Das Utpal***

Abstract

Radial line stubs have been found most suitable in planer microwave circuit configurations to meet the requirement of low impedance particularly at higher frequencies when the width of the conventional straight stub becomes significant fraction of the wavelength. Analysis of radial line stub in microstrip and suspended substrate environments based on variational approach in conjunction with cascaded network theory is presented. Simulated results for radial line stubs reveal that the reactance variation around zero reactance frequency is seen to be slower in the case of suspended substrate radial stubs than that in its microstrip counterpart. It confirms broad band behavior of suspended substrate radial line stub. It is observed that as the angle of stub is increased (60°, 90°, 120°) the flatness of reactance variation is less with frequency near zero impedance, both in the microstrip and the suspended substrate cases. Increasing the length of the stub was not found to be suitable from larger bandwidth considerations. The simulated results for the microstrip butterfly radial line stub are found in close agreement with Giannini's theory and experiment. Both simulation and experimental results have been obtained for suspended substrate radial line stub (both single and butterfly configurations) for operation at 8 - 10 GHz. The theoretical simulations reasonably agree with the experimental data. Experimental data in this work on suspended-substrate-uneven-butteifly-radial-line-stub are also compared with the corresponding simulated results. It is found that the nature of frequency response remain the same in both of the cases but the agreement is better for the symmetric butterfly configuration. The peak rejection of a line with parallel stub shows a maximum of 50 -60 dB of rejection at the resonant frequency. The bandwidths as required in communication network, at 20 dB rejection is found to be approximately 2.2 GHz and 4.0 GHz for single stubs and uneven butterfly radial stub respectively at a radial angle of 60 - 90 in suspended substrate configuration on 0.254 mm thick Duroid 6010.2 substrate (£,- = 10.2). Theoretical simulation for double uneven butterfly stubs connected line can yield a 20 dB rejection bandwidths as high as 8 GHz around a center frequency of 12 GHz. This is suitable for broadband optoelectronics. Simulated data for butterfly radial line stub in broadside coupled suspended substrate line shows that a wider bandwidth is available for the odd mode compared to that of even mode operation. It is observed that for a fixed length of the stub in the cases of microstrip and odd mode broad side coupled lines the frequency separation between the successive resonance freqencies (where the reactance goes to °°) decreases with increasing the stub angle due to variation in the radial length. The reverse is, however, observed for suspended substrate and even mode broad side coupled configurations with increasing the stub angle. Some applications of radial line stub are also pointed out with a remark on first level development of bias-T for opto-electronics applications.

For more details click here **Back**

Title	***:***	***Synchronous Link Converter And Its Application To Electric Traction***
Author(s)	***:***	***Frederick L***
Roll No	***:***	***9610418***
Supervisor(s)	***:***	***Dubey G K***

Abstract

The present work formulates the method for single phase synchronous link converter, which is widely used in modern ac motor traction drive systems, low power industrial ac motor drive systems, UPS systems and var compensators. The experimental results have also been presented. The thesis reviews the modern ac motor traction drive powered from 25 kV, 50 hz single phase system, which has become state of art technology for locomotives / motor coaches. The simulation results to investigate the performance o f synchronous link converter when used as a front end converter in modern regenerative ac motor, motor coach has been presented

For more details click here **Back**

Title : ***Fuzzy Logic Controllers For Load Frequency Control***

Author(s) : ***Rishika***
Roll No : ***9610446***
Supervisor(s) : ***Kalra Prem Kumar***

Abstract

The concept of Fuzzy logic to control a nonlinear process has received wide attention in the recent time. A single generator unit feeds a power line to various users whose power demand can vary over time, is a nonlinear process. If there is a change in load demand then there is a change in the frequency of the generator. This thesis presents procedure to design controllers based on fuzzy logic to control the load frequency variations in an efficient manner. Three types of plants are studied — single area Thermal plant, Two area Thermal -Thermal plant and Two area Hydro -Thermal plant. Firstly, fuzzy gain scheduling controllers are designed which change the controller's parameters according to the variations. The results show the superiority of such contro llers. In the second case, fuzzy logic controllers are designed. Each controller shows the characteristics of its linear controller. However, the results obtained with fuzzy logic controllers are not optimized.

For more details click here **Back**

Title : ***Analysis And Controller Design By Pole Placement For A Dc-Dc Buck - Boost Converter***

Author(s) : ***Sachdev P P***
Roll No : ***9610434***
Supervisor(s) : ***Joshi Avinash***

Abstract

A detailed analysis of modelling and behaviour of buck boost converter has been done. The state space linear differential models are used to study the open loop time domain behaviour of the converter. To enable design of close loop feed back controllers co ntinuous time representation of the converter is achieved by small signal linearized model which are obtained by linearizing the non linear state space averaged equations. Attempt is made to high light the fact that if switching information is neglected t hen state space average model can represent the converter behaviour for all kinds of perturbation. Using the small signal linearized model a state feed back controller is designed by pole placement and its response for perturbation in V ref , V dc , R has bee n simulate.

For more details click here **Back**

Title **:** ***Modal Analysis Of Voltage Stability Of MTDC-AC Systems***
Author(s) **:** ***Reddy Gaddam Madhukar***
Roll No **:** ***9610411***
Supervisor(s) **:** ***Prabhu S S***

Abstract

The thesis work involves instantaneous reactive and harmonic power compensation of a three - phase system. A typical controlled rectifier load is considered to create harmonics and phase displacement for the source current. Using the concept of instantaneous power the compensator system performs well both in steady state and transient state as well. The main intention is to operate the voltage source inverter at Constant Switching Frequency with relatively a low value of inductor. Because of a small inductance, the response of compensator system becomes faster. Due to constant switching frequency technique the system becomes more compact and reduces switching losses and device stresses. The voltage source inverters studied, include two - level, three - level and five level inverters. Higher level inverters increase the complexity ad need voltage control even for a lossless system.

For more details click here **Back**

Title : *Comparison Of Different Modeling Techniques*
Author(s) : *Chandra V Sharat*
Roll No : *9610462*
Supervisor(s) : *Kalra Prem Kumar*

Abstract

The method of modeling studied in the present work include fuzzy lest square regression, cluster wise fuzzy regression fuzzy auto regressive moving average (ARMA) method, Sugeno - type fuzzy system identification technique and orthogonal parameter estimator embedded with fuzzy discrimination (ortho - clustering) technique. Fuzzy least square regression with its forvaritiuons Are developed. The effect of different modeling functions upon the performance of the fuzzy regression is evaluated. Cluster wise regressi on is used mainly to deal with the heterogeneity of the observed data. The effe3ct of different clustering algorithms upon the performance of the cluster regression is evaluated. The simple fuzzy regression method is applied to the ARAM technique for model ing a dynamical system. Sugenu - type fuzzy identification technique for modeling a dynamical system. Sugeon - type fuzzy identification technique is developed to include the fuzzy reasoning and implications in modeling of the system. The premise parameters an d consequence parameters identification is separated through ortho - clustering technique and the effect of different clustering methods on its performance is evaluated. The above algorithms are applie3d to the problems of estimation of life converter lining and modeling of Box Jenkins furnace for evaluating their performance .A comparison of performance of all the developed methods brought out from the results of the two test problems. The test problems are also modeled through neural networks and results ar e presented for comparison purpose. The performance of cluster wise fuzzy regression is the best of all the other techniques for modeling a system having inherent imprecision and/or having very few data describe the system, where as for a simple system lik e Box Jenkins' gas furnace cluster wise conventional regression has better performance

For more details click here **Back**

Title : ***Design Of Neural Network Controllers For Load Frequency Control***

Author(s) : ***Shrikhande Rashmi***

Roll No : ***9610444***

Supervisor(s) : ***Kalra Prem Kumar***

Abstract

The problem of control design for complex nonlinear systems is not sufficiently addressed in the conventional control techniques. The field of intelligent control was developed for solving such problems. This thesis describes an application of layered neural networks to load frequency control in power systems. An attempt has been made to evaluate the performance of such neural network controllers when implemented in single area and two data interconnected power systems. Two approaches have been proposed . In the first approach (a supervised controller) controller is trained for a given value of disturbance. Cumulative error minimization has been carried out to incorporate the past histories. In the second approach (a self learning controller), the back pr opagation of error through the plant is carried out. This effectively minimizes the overall plant error. The simulation results reveal that both types of controllers are able to track the optimal response of the conventional (integral) controller under cer tain specific conditions. For its general applicability to real control situations, some important problem are required to be addressed and more theoretical results need to be studied

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For more details click here **Back**

Title : *Dimensionality Reduction Tchniques For Intelligent Systems*
Author(s) : *Shrivastava Manish Kumar*
Roll No : *9610423*
Supervisor(s) : *Kalra Prem Kumar*

Abstract

The demand for the large number of instances grows exponentially with dimensionality of feature space. In addition, greater computational complexity is implied by higher dimensionality. Various modeling techniques face such problems. So before inputting the samples or instances for modeling. It is always prederred to preprocess the available data set, which include data prioritization and feature clustering. Principal component analysis (PCA) and independent component analysis (ICA) are two commonly used data prioritization techniques, which absorb most of data variation with smaller dimension. In PCA< we have uncorrelated components and in ICA, we have independent component. Heterogeneity is always present in the data, which also offers problems in modeling. Clustering or grouping of samples (generally features based) is an answer to such problem. Kmeans clustering, partitions the total data set in to k classes. Sometimes, in K-means clustering, there may be some classes which are empty. Such problem is taken care in fuzzy C-means clustering, in which each samples belongs to all c classes with some membership. A class of neural networks also performs clustering and classification. Kohenon self organizing map performs similar action to K-means method. But, two unsupervised networks ART2 and Fuzzy ART classify the samples depending up on a factor called vigilance factor. The advantage of these networks is that, we need not to specify the number of classes in advance. Inductive reasoning (ID3) can also be applied for this purpose. All these techniques have been applied to steel converter lining life prediction problem. Using PCA and ICA, we could reduce the dimension of the system to 15x13 from 15x26. then these 15x13 system are given as input to various modeling techniques. Generally, ICA is giving good results. In cluster wise modeling, K-means Fuzzy C-means and kohenon clustering give better results

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For more details click here **Back**

Title : ***Comparision Of Various Layer Structured Modelling Techniques***

Author(s) : ***Biswal Neelamadhab***
Roll No : ***9610431***
Supervisor(s) : ***Kalra Prem Kumar***

Abstract

This thesis work includes various system modeling technique based on layer structure. Different techniques applied are artificial neural networks (ANN), group method of data handling (GMDH) and fuzzy group method of data handling (FGMDH). In ANN back propagation algorithm is used for input and output mapping. All these modeling techniques are applied to the data available from research and development cell of SAIL. The aim behind the application of these system modeling techniques are the prediction of life of the converter lining and subsequent sensitivity analysis to find out important parameters. Pre - processing of these data are performed through independent component analysis (ICA) and principal component analysis (PCA). From preprocessing of data, required number of input parameters are selected from all available inputs. These data sets are used for system modeling. Training samples are trained through GMDH fuzzy GMDH and ANN, and life of the converter lining is predicted for prediction samples. Er ror analysis is performed for all these system modeling techniques. In GMDH and fuzzy GMDH various functions i.e. sin, tan - hyperbolic, sigmoidal etc and polynomials are used for prediction purpose. Sensitivity analysis is made through above techniques. Sen sitivity analysis includes selection of most effective parameters and their effect (positive or negative). For small charges in inputs, variation in corresponding outputs are predicted. This analysis, can be used to increase the life of the converter linin g by adjusting different input parameters. Ultimately results of various functions used in GMDH FMGDH and ANN are compared

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For more details click here **Back**

Title : *A Novel Control Strategy For Mitigation Of SSR Employing Svc Auxiliary Controllers*

Author(s) : *Gupta Yachika*
Roll No : *9610468*
Supervisor(s) : *Varma Rajiv K*

Abstract

Static var compensators (SVC) are traditionally placed at the generator terminals to mitigate subsynchronous oscillations (SSO) in series compensated electric power transmission systems. An auxiliary control [14] of the SVC employing generator rotor fr equency feedback is shown to be adequate for damping SSO. It has been reported [15] that an SVC located at the midpoint of a series compensated line can be utilized for dual purposes of damping SSO and stability enhancement. Controllers of these SVCs have been designed based on a combination of line current signal and computed value of generator internal frequency (CIF). Both these signals utilize local measurements, as the current trend in power system practice is to use local signals for the sake of reliability. In this thesis an altogether new concept of SVC control employing a remote generator frequency signal transmitted over a telecom line, is presented for damping of SSO in series compensated transmission systems. The IEEE First SSR Bench mark system is suitably modified to include an SVC at the midpoint of transmission line. Auxiliary controllers are designed for various signals such as line current, computed internal frequency (CIF) and remote frequency signal. Their effectiveness is then investigated for damping all the torsional modes at all critical levels of series compensation. Eigenvalue analysis is utilized to examine the system stability in each case. It is concluded in this thesis that this remote rotor frequency signal toge ther with line current signal can be successfully applied for damping all the torsional modes at all the critical levels of series compensation. The performance of remote rotor frequency signal is much better than the computed internal frequency signal. Th is remote frequency signal is observed to be efficient over a reasonable range of telecom delays. This concept of remotely transmitted signal is being investigated for the first time in the control of static var compensators to damp subsynchronous osci llations

For more details click here **Back**

Title : ***Three-Phase Instantaneous Reactive And Harmonic Power Compensation Using Multi-Level Inverters At Constant Switching Frequency***

Author(s) : ***Bhujabal Brajendra Kumar***

Roll No : ***9610407***

Supervisor(s) : ***Doradla S R***

Abstract

The thesis work involves instantaneous reactive and harmonic power compensation of a three - phase system. A typical controlled rectifier load is considered to create harmonics and phase displacement for the source current. Using the concept of instantaneous power the compensator system performs well both in steady state and transient state as well. The main intention is to operate the voltage source inverter at Constant Switching Frequency with relatively a low value of inductor. Because of a small inductanc e, the response of compensator system becomes faster. Due to constant switching frequency technique the system becomes more compact and reduces switching losses and device stresses. The voltage source inverters studied, include two - level, three - level and five level inverters. Higher level inverters increase the complexity ad need voltage control even for a lossless system. A three - phase three - level inverters, consisting of three wye - connected single - phase inverters, is studied for the compensator system. I t has been found that this configuration needs no additional voltage control aspect for a lossless system. This three -level inverter is operated at constant switching frequency to obtain instantaneous reactive and harmonic power compensation. System loses are included and the capacitor voltages are held constant with an outer voltage control loop. The study reveals that the compensator employing wye - connected three -phase three level inverter provides independent voltage and current control capability with a ll the advantages associated with constant switching frequency

For more details click here **Back**

Title : ***Coordinated Control Of Facts Devices In Electrical Power Systems***

Author(s) : ***Rao Madduri Srinivasa***
Roll No : ***9610419***
Supervisor(s) : ***Varma Rajiv K***

Abstract

This thesis presents small signal stability evaluation for a single machine infinite bus system with two parallel lines. O ne line is series compensated by a Thyristor Controlled Series Capacitor (TCSC) and the other line is shunt compensated at the mid point by a Static Var Compensator (SVC). E igenvalue studies are performed to explore the possibilities of increasing the power transfer capability of different combinations of the study system. P ower System Stabilizer (PSS) when added to the excitation sys tem damps the system oscillations for all the investigated system configurations. TCSC when employed with Constant Current (CC) controller provides good system damping compared to TCSC with Constant Angle (CA) controller. I t is shown that a combination of TCSC with CC controller, SVC and PSS results in a significant increase of power transfer capability of the system. I n addition such combination of FACTS devices ensures improved steady state voltage profile and simultaneous control of power sharing between two parallel lines of the study system.

For more details click here **Back**

Title : *Load Modelling For Power Flow And Stability Studies*
Author(s) : *Rajan T P Varada*
Roll No : *9610459*
Supervisor(s) : *PrabhuS S*

Abstract

This work incorporate lode characteristics in power flow and stability studies. Acomputer code has been developed for modeling various kinds of loads. It gives real andreactive power of different loads for specified voltage and frequency. This code has beenlinked with a power flow program; the composite program with some addition facilities isnamed “LOADMOD”. LOADMOD consists of two main parts, namely, Load Flow Part (LFP) LoadModelling Part (LMP). In LMP, facility is given to user to define a number of models,their parameters choice of buses for load modeling to be applied, and distribution of loadamong defined models. Compensation at the load buses is also modeled. Real andreactive power losses in the feeders have been modeled, for which the initial real and thereactive power losses at the reference voltage and frequencies are needed. LFP interactswith LMP, i.e., LFP gives voltage magnitudes at different buses to LMP, LMP calculatesP and Q and gives them as input to LFP and so on till voltages converge. LOADMOD has been tested on NREB (357) and IEEE (30BUS) systems with someassumed load mix at some chosen buses. A substantial change in the final voltages wasobserved with respect to the results for direct specification of real and reactive powers atload buses. Facility has been provided in LOADMOD to iteratively increase the loadingby a fixed factor at specified or all load buses in the system, at the end of each LFP+LMPrun till LFP fails to coverage. This failure to coverage indicates that the system hasreached the static stability limit. Changes in the voltage have been studied with increasingloading for the IEEE 30 bus system in order to capture the buses prone to voltageinstability. Facility has been provided in LOADMOD for load and generator outages when theloading on the system is increased. The results for such contingencies in IEEE 30 bussystem have been studied

For more details click here **Back**

Title **:** ***Breakdown Properties Of Atmospheric Air With Switching Overvoltages***

Author(s) **:** ***Singh Surendra***
Roll No **:** ***9610458***
Supervisor(s) **:** ***Arora Ravindra***

Abstract

Dielectrics, the indispensible part of modern power systems are subjected to severe stress of lightning and switching impulsess. The dielectric used should be able to with stand the severest of the impulses. In this work four spheiesphere configurations of radius 20 mm. 15 mm, 12.5 mm, 7.5 mm are used. The gap distance is varied from 2 cm to 15 cm making the field to vary from weakly non uniform to non uniform field. This study analyses the variation of breakdown strength and breakdown voltage under different switching impulses. It is also studied how breakdown strength of dielectric changes under different field configurations. How the polarity effect under different field configurations affects the breakdown strength is also studied. The propagation time and propagation velocity (in cm//is) of streamer in the dielectrics are also measured. Their changes under different field con figurations are also recorded

For more details click here **Back**

Title	***:***	***A New Fourth Order Resonant Dc-Dc Converter With PWM Control***
Author(s)	***:***	***Tripathi Gyandendra Dev***
Roll No	***:***	***9610413***
Supervisor(s)	***:***	***Doradla S R***

Abstract

Many of the limitation of two element resonant topologies can be over come by adding the third reactive element. However, the number of possible circuit topologies increase as number of reactive elements increase. The selection of a proper higher order topology for a given application is a difficult task. In this work a new LC - LC type resonant converter using a capacitor output filter and providing load independent operation at two frequencies is presented. Pulse width modulation (PWM) is employed to control the output voltage. The possible modes of converter operation under PWM are analyzed using state space equations with numerical solution approach. A new analysis technique by state variable tr ansformation and developing decoupled state space equation is developed for frequency controlled converter. This method turns out to be an efficient analysis tool for higher order converters. Steady state solutions are obtained by running the simulations f or sufficiently large times. Closed form solutions are worked out based on a simple analysis technique - Complex circuit analysis technique. An extensive simulation has been carried out using a C Program which is compared with SPICE software package for ver ification. The C program was used extensively to study the transient and steady state performance. There is complete agreement between the results obtained by the above program and those obtained by SPICE.

For more details click here **Back**

Title	***:***	***Neural Network Algorithms For Character Recognition***
Author(s)	***:***	***Kumari Pratibha***
Roll No	***:***	***9710439***
Supervisor(s)	***:***	***Kalra Prem Kumar***

Abstract

Artificial Neural Network (ANN) mimics the computational behavior of living cellular system, which acquires and stores knowledge. It has demonstrated its capability for solving complex pattern recognition problems. This thesis presents ANN algorithms and their implementation for the recognition of the 26 English alphabets. A thorough analysis and implementation of nine algorithms- Bidirectional Associative Memory, Hopfield Autoassociative Memory, Feature Recognition Neural Network, Hamming Network and MAXNET, Backpropogation algorithm, Quickprop, Adaptive Resonance Theory 1, Kohonen Self Oraganising Map and Neocognitron, has been carried out . The relative performance, the merits and the demerits of each algorithm has been studied. These algorithms are compared on the basis of the number of neurons and the number of unknowns to be computed. The comparison of the algorithms has also been performed based on criteria. - Noise in weights, Noise in inputs, Loss of connections, Missing informations and, Adding informations. It has been observed that a certain algorithm performs best under a particular criterion

For more details click here **Back**